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UTILITY PATENT APPLICATION TRANSMITTAL
(Large Entity)
(Only for new nonprovisional applications under 37 CFR 1.53(b))

Docket No.
FUR0007-US

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Box Patent Application
Washington, D.C. 20231

Transmitted herewith for filing under 35 U.S.C. 111(a) and 37 C.F.R. 1.53(b) is a new utility patent application for an invention entitled:

AUDIO SIGNAL PROCESSING CIRCUIT

and invented by:

JOJI KASAI, KAZUMASA TAKEMURA AND TETSURO NAKATAKE

If a CONTINUATION APPLICATION, *check appropriate box and supply the requisite information:*

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Application Elements

1. ☒ Filing fee as calculated and transmitted as described below
2. ☒ Specification having 46 pages and including the following:
   - a. ☒ Descriptive Title of the Invention
   - b. ☐ Cross References to Related Applications *(if applicable)*
   - c. ☐ Statement Regarding Federally-sponsored Research/Development *(if applicable)*
   - d. ☐ Reference to Microfiche Appendix *(if applicable)*
   - e. ☒ Background of the Invention
   - f. ☒ Brief Summary of the Invention
   - g. ☒ Brief Description of the Drawings *(if drawings filed)*
   - h. ☒ Detailed Description
   - i. ☒ Claim(s) as Classified Below
   - j. ☒ Abstract of the Disclosure

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Application Elements (Continued)

3. ☒ Drawing(s) *(when necessary as prescribed by 35 USC 113)*
   - a. ☒ Formal — Number of Sheets **Twenty-Nine (29) Only**
   - b. ☐ Informal — Number of Sheets ______
4. ☒ Oath or Declaration
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   - b. ☐ Copy from a prior application (37 CFR 1.63(d)) *(for continuation/divisional application only)*
   - c. ☐ With Power of Attorney ☐ Without Power of Attorney
   - d. ☐ *DELETION OF INVENTOR(S)*
     Signed statement attached deleting inventor(s) named in the prior application, see 37 C.F.R. 1.63(d)(2) and 1.33(b).
5. ☐ Incorporation By Reference *(usable if Box 4b is checked)*
   The entire disclosure of the prior application, from which a copy of the oath or declaration is supplied under Box 4b, is considered as being part of the disclosure of the accompanying application and is hereby incorporated by reference therein.
6. ☐ Computer Program in Microfiche *(Appendix)*
7. ☐ Nucleotide and/or Amino Acid Sequence Submission *(if applicable, all must be included)*
   - a. ☐ Paper Copy
   - b. ☐ Computer Readable Copy *(identical to computer copy)*
   - c. ☐ Statement Verifying Identical Paper and Computer Readable Copy

Accompanying Application Parts

8. ☐ Assignment Papers *(cover sheet & document(s))*
9. ☐ 37 CFR 3.73(B) Statement *(when there is an assignee)*
10. ☐ English Translation Document *(if applicable)*
11. ☐ Information Disclosure Statement/PTO-1449 ☐ Copies of IDS Citations
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15. ☐ Certified Copy of Priority Document(s) *(if foreign priority is claimed)*

16. ☐ Additional Enclosures *(please identify below):*

Fee Calculation and Transmittal

CLAIMS AS FILED

For	#Filed	#Allowed	#Extra		Rate	Fee
Total Claims	16	- 20 =	0	x	$18.00	$0.00
Indep. Claims	8	- 3 =	5	x	$78.00	$390.00
Multiple Dependent Claims (check if applicable) ☐						$0.00
					BASIC FEE	$760.00
OTHER FEE *(specify purpose)*						$0.00
					TOTAL FILING FEE	$1,150.00

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Signature
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Regn. No. 32,329

Dated: **JULY 28, 1999**

cc:

P01ULRG/REV04

AUDIO SIGNAL PROCESSING CIRCUIT

Cross-Reference to Related Applications

The disclosure of Japanese Patent Application Nos. Hei 10-217929 and Hei 10-218128 both filed on July 31, 1998 including specification, claims, drawings and summary is herein incorporated by reference in its entirety.

BACKGROUND OF THE INVENTION

1. FIELD OF THE INVENTION

The present invention relates to an audio signal processing circuit in a so-called surround system. More particularly, the present invention relates to simplification of its structure, improvement of accuracy, and localization of sound image.

2. DESCRIPTION OF THE RELATED ART

Recently, an audio reproduction apparatus having surround channels at a left and a right sides to a listener in addition to a left and a right (and optionally a center) front channels, has been developed not only for business use but also for home use. In the surround reproduction utilizing

such apparatus, two of surround speakers are usually arranged at the both sides (i.e., left and right sides) to the listener. When the correlation between the left and the right surround signals is small (i.e., when a stereophonic surround system is employed), the listener does not have an unnatural feeling. In contrast, when the correlation between the left and the right surround signals is large (i.e., when a monophonic surround system is employed), the following problem is recognized depending on the listener's position. Specifically, when the listener is positioned at the center between the left and the right surround speakers, the listener has an unnatural feeling as if sound image was localized in the head of the listener.

In order to solve the above-mentioned problem, a technique alternatively dividing a monophonic signal into two channels with respect to each frequency component of predetermined width by using a comb type filter so as to virtually reproduce stereophonic sound, a technique performing a pitch shift processing so as to reduce the correlation (e.g., THX system), and a technique performing a 90 degrees phase shift processing so as to make the correlation zero, have been proposed.

However, the above-mentioned techniques have the following problems, respectively.

According to the technique using the comb type filter so as to virtually reproduce stereophonic sound, unnaturally large sound is often reproduced when a musical instrument is used as sound source. Furthermore, the virtual stereophonic sound reproduction compromises the sound quality when the surround signals are stereophonic. Therefore, it is necessary to prevent the stereophonic sound reproduction in such a case. As a result, a change of a processing mode is required depending upon whether the surround signals are monophonic or stereophonic, which makes the overall processing complicated.

According to the technique performing the pitch shift processing such as THX system, there has been a tradeoff problem that the large amount of the pitch shift is required for reducing the correlation and that the large amount of the pitch shift lowers the sound quality. Furthermore, similar to the virtual stereophonic sound reproduction, a change of a processing mode is required depending upon whether the surround signals are monophonic or stereophonic, which makes the overall processing complicated.

The technique performing the 90 degrees phase shift processing is superior to the above-described techniques in view of the fact that the sound quality is not lowered in the

case of the stereophonic surround signals and that a change of a processing mode is not required. However, sound image is apt to be localized in the direction of the channel whose phase relatively progresses, which provides the listener with an unnatural feeling. This problem is especially remarkable in the case where the left and the right surround sound sources are virtual sound sources.

As described above, an apparatus and a method, which are capable of performing the same processing independent of whether the surround signals are monophonic or stereophonic, preventing sound image localization in the head of the listener so as to create sound field just as enveloping the listener, and performing a processing which does not compromise the sound quality even when the surround signals are stereophonic, are eagerly demanded.

By the way, an audio signal processing circuit disclosed in Japanese Laid-open Publication No. Hei 8-265899 (265899/1996) is shown in Figure **29**. The circuit is used for making a listener **102** to feel that sound image reproduced by virtual speakers **XL** and **XR** is virtually localized at rear sides to the listener **102**. By utilizing the circuit, the listener is able to feel that he/she is surrounded by the sound reproduced with the speakers **104L** and **104R** as well as surrounded by the sound reproduced with the virtual speakers

XL and **XR** even when the speakers **104L** and **104R** are actually arranged only in front of the listener **102**.

In the apparatus shown in Figure **29**, a total of four filters **106a, 106b, 106c** and **106d** are used for performing the above-mentioned sound image localization. Transfer functions H11, H12, H21 and H22 of the respective filters are represented by the following equations:

$$H11=(h_{RR}h_{L'L}-h_{RL}h_{L'R})/(h_{LL}h_{RR}-h_{LR}h_{RL})$$
$$H12=(h_{LL}h_{L'R}-h_{LR}h_{L'L})/(h_{LL}h_{RR}-h_{LR}h_{RL})$$
$$H21=(h_{RR}h_{R'L}-h_{RL}h_{R'R})/(h_{LL}h_{RR}-h_{LR}h_{RL})$$
$$H22=(h_{LL}h_{R'R}-h_{LR}h_{R'L})/(h_{LL}h_{RR}-h_{LR}h_{RL})$$

Here, h_{LL} is a transfer function from the speaker **104L** to the left ear **102L** of the listener **102**, h_{LR} is a transfer function from the speaker **104L** to the right ear **102R** of the listener **102**, h_{RL} is a transfer function from the speaker **104R** to the left ear **102L** of the listener **102**, and h_{RR} is a transfer function from the speaker **104R** to the right ear **102R** of the listener **102**.

Equations $h_{LL}=h_{RR}$, $h_{LR}=h_{RL}$, $h_{L'L}=h_{R'R}$ and $h_{L'R}=h_{R'L}$ are satisfied in the equations stated above when the speakers **104L** and **104R** and the virtual speakers **XL** and **XR** are symmetrically arranged with respect to a central axis **108** through the

listener **102**. As a result, equations H11=H22 and H12=H21 can be derived, so that the circuit can be obtained by utilizing total of two filters as shown in Figure **30** (such structure is referred to as "shuffler type filter"). Here, transfer functions H_{SUM} of the filters **110a** and H_{DIF} of the filters **110b** are represented by the following equations:

$$H_{SUM}=(ha'+hb')/2(ha+hb)$$
$$H_{DIF}=(ha'-hb')/2(ha-hb)$$

wherein equations $ha=h_{LL}=h_{RR}$, $hb=h_{LR}=h_{RL}$, $ha'=h_{L'L}=h_{R'R}$ and $hb'=h_{L'R}=h_{R'L}$ are satisfied.

As described above, in the case where the speakers are symmetrically arranged, sound image can be localized at the virtual speaker positions with the simple circuit.

Furthermore, a method for localizing sound image by utilizing a cross-feed filter **112** and a cross-talk cancel filter **114** as shown in Figure **31**, has been proposed. The cross-talk cancel filter **114** functions to cancel cross-talk from the right speaker **104R** to the left ear **102L** of the listener and that from the left speaker **104L** to the right ear **102R** of the listener. Accordingly, the cross-talk cancel filter **114** makes it possible that a left channel signal **L** reaches only the left ear **102L** and a right channel signal **R** reaches only the right ear **102R**. As a result, sound image

can be localized at the desired position by adjusting the amount of the cross-talk with the cross-talk cancel filter **114**.

The above-mentioned cross-talk cancel filter **114** can also be obtained by utilizing the shuffler type filter as shown in Figure **30**. In this case, transfer functions H_{SUM} of the filters **110a** and H_{DIF} of the filters **110b** are represented by the following equations:

$$H_{SUM}=ha/(2(ha+hb))$$

$$H_{DIF}=ha/(2(ha-hb)).$$

According to the shuffler type filter, a circuit having satisfactory sound image localization ability or satisfactory cross-talk cancel ability can be obtained only when the filters **110a** and **110b** are highly accurate. However, in order to make the filters accurate, the structure thereof becomes complicated. As a result, when a digital signal processor (DSP) is employed for the filters, it takes much time to perform a sound image localization processing or a cross-talk cancel processing. In contrast, when the structure of the filters is simple, the ability of the filters is insufficient.

As described above, a shuffler type filter having a simple structure and a high accuracy is eagerly demanded for a surround system.

SUMMARY OF THE INVENTION

An audio signal processing circuit according to the present invention is used for an audio reproduction apparatus at least having sound source located substantially at left and right sides to a listener. The audio signal processing circuit includes a phase difference control portion. The phase difference control portion receives a left channel signal for the left sound source and a right channel signal for the right sound source, controls a phase difference between the left and right channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference controlled left and right channel signals for the left and right sound source, respectively.

The phase difference of 60 degrees causes the problem that sound image is localized in the direction of the channel whose phase relatively progresses, as in the case of the 90 degrees phase shift processing. The phase difference of 180 degrees (i.e., inverse phase) causes a listener unpleasant feeling as if the ear of the listener is pressurized, which

problem is unique to the inverse phase. In contrast, the phase difference of 140 to 160 degrees does not cause an unpleasant feeling unique to the inverse phase or produces sound image localization in the certain direction. As a result, the present invention can prevent sound image of the monophonic signal from localizing in the head of the listener so as to create sound field just as enveloping the listener.

Furthermore, since only the phase difference control operation is additionally performed according to the present invention, the audio reproduction according to the present invention does not compromise the sound quality even when the stereophonic signal is employed. As a result, according to the present invention, the same processing can be performed independent of whether the input signal is monophonic or stereophonic.

In one embodiment of the invention, the phase difference control portion produces the relative phase difference of 140 degrees to 160 degrees in a frequency region ranging from 200 Hz to 1 kHz. Accordingly, the phase difference control can be effectively performed while the structure of the phase difference control portion is made simple.

According to another aspect of the present invention, a surround audio reproduction apparatus having a left and a

right channels in front of a listener and a left and a right surround channels at left and right sides with respect to the listener, is provided. The apparatus includes a phase difference control portion. The phase difference control portion receives a left surround channel signal and a right surround channel signal, controls a phase difference between the left and the right surround channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference controlled surround left and right channel signals for a left and a right surround sound source, respectively. Accordingly, an audio reproduction apparatus capable of performing the same processing independent of whether the input signals are monophonic or stereophonic, preventing sound image localization in the head of the listener so as to create sound field just as enveloping the listener, and performing a processing which does not compromise the sound quality even when the surround signals are stereophonic, can be obtained.

In one embodiment of the invention, the left and the right surround sound sources are a virtual sound source produced by a sound image localization processing.

In another embodiment of the invention, the phase difference control portion produces the relative phase

difference of 140 degrees to 160 degrees in a frequency region ranging from 200 Hz to 1 kHz. Accordingly, the phase difference control can be effectively performed while the structure of the phase difference control portion is made simple.

According to another aspect of the present invention, an audio reproduction method at least utilizing sound source located substantially at left and right sides to a listener, is provided. The method includes the steps of: controlling a phase difference between a left channel signal for the left sound source and a right channel signal for the right sound source so as to produce a relative phase difference in the range of 140 degrees to 160 degrees; and outputting the phase difference controlled left and right channel signals for the left and right sound source, respectively.

According to still another aspect of the present invention, a shuffler type audio signal processing circuit is provided. The shuffler type audio signal processing circuit includes a first filter for producing a sum signal of a left channel signal and a right channel signal; and a second filter for producing a differential signal of the left channel signal and the right channel signal. In a shuffler type audio signal processing circuit, a gain of the second filter is higher than that of the first filter in a low

frequency region. Accordingly, by making an accuracy of the second filter higher than that of the first filter in a low frequency region, the structure of the circuit can be simplified while a reduction of accuracy is prevented.

According to still another aspect of the present invention, a shuffler type audio signal processing circuit is provided. The shuffler type audio signal processing circuit includes a first filter for producing a sum signal of a left channel signal and a right channel signal; and a second filter for producing a differential signal of the left channel signal and the right channel signal, wherein the first filter and the second filter are FIR filter, and the tap number of the second filter is larger than that of the first filter. Accordingly, the structure of the circuit can be simplified while a reduction of accuracy is prevented.

In one embodiment of the invention, the second filter is composed of a filter bank. Accordingly, a processing margin can be increased by performing down-sampling.

In another embodiment of the invention, the filter bank performs down-sampling by the larger number for the lower frequency component. Accordingly, an accuracy of the second filter is made higher than that of the first filter in a low

frequency region, so that the structure of the circuit can be simplified while a reduction of accuracy is prevented.

According to still another aspect of the present invention, a shuffler type audio signal processing circuit is provided. The shuffler type audio signal processing circuit includes a first filter for producing a sum signal of a left channel signal and a right channel signal; and a second filter for producing a differential signal of the left channel signal and the right channel signal, wherein the first filter is FIR filter and the second filter is composed of a parallel connection of FIR filter and secondary IIR filter. Accordingly, an accuracy of the second filter is made higher than that of the first filter in a low frequency region, so that the structure of the circuit can be simplified while a reduction of accuracy is prevented. Furthermore, since a low frequency component can be processed with the secondary IIR filter, unnecessary increase of the tap number of the FIR filter can be prevented.

In one embodiment of the invention, the second filter includes: FIR filter, and secondary IIR filter connected in parallel to the FIR filter at one of the intermediate taps or the end tap thereof. Accordingly, an accuracy of the second filter is made higher than that of the first filter in a low frequency region, so that the structure of the circuit can

be simplified while a reduction of accuracy is prevented. Furthermore, by varying an intermediate tap connected to the secondary IIR filter, optimum properties for the filter can be obtained.

In one embodiment of the invention, the circuit is used as a cross-talk cancel filter.

In one embodiment of the invention, the circuit is used as a sound image localization processing filter.

According to still another aspect of the present invention, a filter is provided. The filter includes: FIR filter having a plurality of taps, IIR filter whose input is connected to one of the intermediate taps or the end tap of the FIR filter, and an adding means which adds outputs of the FIR filter and the IIR filter. Accordingly, a filter having desired properties can be obtained.

According to still another aspect of the present invention, a shuffler type audio signal processing method is provided. The method includes the steps of: performing a first filtering process for a sum signal of a left channel signal and a right channel signal; and performing a second filtering process for a differential signal of the left channel signal and the right channel signal, wherein an

accuracy of the second filtering process is higher than that of the first filtering process.

Thus, the invention described herein makes the possible the advantages of: (1) providing a processingcapable of performing the same processing independent of whether the input signals are monophonic or stereophonic, preventing sound image localization in the head of the listener so as to create sound field just as enveloping the listener, and performing a processing which does not compromise the sound quality even when the surround signals are stereophonic; and (2) providing a shuffler type filter having a simple structure and a high accuracy.

These and other advantages of the present invention will become apparent to those skilled in the art upon reading and understanding the following detailed description with reference to the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure **1** is a block diagram of an audio signal processing circuit according to an embodiment of the present invention.

Figure **2** is a block diagram of an audio reproduction apparatus wherein the audio signal processing circuit of Figure **1** is incorporated.

Figures **3A** and **3B** are circuit diagrams according to embodiments wherein an all pass filter used in the present invention is composed of an analog circuit.

Figure **4** is a graph illustrating a frequency-phase relationship of the all pass filter used in the present invention.

Figure **5** is a schematic view illustrating an arrangement of speakers in accordance with a surround audio reproduction apparatus of the present invention.

Figure **6** is a block diagram according to an embodiment wherein the audio signal processing circuit of the present invention is applied to a surround audio reproduction apparatus which produces virtual sound sources by a sound image localization processing using DSP.

Figure **7** is a schematic view illustrating an example of an arrangement of the virtual sound sources of Figure **6**.

Figure **8** is a signal-flow diagram illustrating the sound image localization processing using DSP.

Figure **9** is a signal-flow diagram illustrating an embodiment wherein an all pass filter used in the present invention is composed of a secondary IIR filter.

Figure **10** is a signal-flow diagram according to another embodiment of the present invention.

Figure **11** is a schematic view illustrating an example of an arrangement of the virtual sound sources of Figure **10**.

Figure **12** is a schematic view of a shuffler type filter according to an embodiment of the present invention.

Figure **13** is a block diagram illustrating a hardware structure of the audio reproduction apparatus using DSP.

Figure **14** is a signal-flow diagram illustrating processings carried out by the DSP in accordance with program(s) stored in a memory.

Figure **15** is a graph illustrating a frequency response H_{SUM} of a first filter and a frequency response H_{DIF} of a second filter, and a cross-talk cancel response Zt1 and a cross-

talk cancel error Zt2 when the first and the second filters are used, wherein both of the first and the second filters have 32 taps.

Figure **16** is a graph illustrating H_{SUM}, H_{DIF}, Zt1 and Zt2 wherein both of the first and the second filters have 64 taps.

Figure **17** is a graph illustrating H_{SUM}, H_{DIF}, Zt1 and Zt2 wherein both of the first and the second filters have 96 taps.

Figure **18** is a graph illustrating H_{SUM}, H_{DIF}, Zt1 and Zt2 wherein the first filter has 32 taps and the second filter has 96 taps.

Figure **19** is a signal-flow diagram according to an embodiment using a filter bank.

Figure **20** is a graph illustrating a cross-talk cancel response Zt1 and a cross-talk cancel error Zt2 when the cross-talk cancel filter shown in Figure **14** is used wherein a first filter having 32 taps and a second filter having 128 taps are incorporated.

Figure **21** is a graph illustrating a cross-talk cancel response Zt1 and a cross-talk cancel error Zt2 when the cross-talk cancel filter shown in Figure **19** is used wherein

a first filter having 32 taps and a second filter corresponding to 128 taps are incorporated.

Figure **22** is a signal-flow diagram according to an embodiment wherein the second filter **120b** is composed of a parallel connection of FIR filter and IIR filter.

Figure **23** is a graph illustrating a frequency response H_{SUM} of the first filter and a frequency response H_{DIF} of the second filter, and a cross-talk cancel response Zt1 and a cross-talk cancel error Zt2 when the cross-talk cancel filter shown in Figure **22** is used.

Figure **24** is a signal-flow diagram according to an embodiment wherein an intermediate tap of FIR filter is connected to an input of IIR filter.

Figure **25** is a graph illustrating a desired impulse response for the second filter.

Figure **26** is a graph illustrating an impulse response of IIR filter having properties approximate to that of Figure **25**.

Figure **27** is a graph illustrating a deviation of the impulse response of the IIR filter from the desired impulse response.

Figure **28** is a graph illustrating an impulse response of FIR filter obtained in due consideration of the deviation of Figure **27**.

Figure **29** is a schematic view illustrating conventional sound image localization technique.

Figure **30** is a circuit diagram illustrating shuffler type filter.

Figure **31** is a block diagram of a sound image localization circuit including a cross-feed filter and a cross-talk cancel filter.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Figure **1** is a block diagram of an audio signal processing circuit according to an embodiment of the present invention. The audio signal processing circuit includes a phase difference control portion **2**. The phase difference control portion **2** receives a left channel signal $\mathbf{S_L}$ for a left sound source $\mathbf{S_{SL}}$ located substantially at a left side to a listener

(shown in Figure **5**) and a right channel signal S_R for a right sound source S_{SR} located substantially at a right side to the listener (also shown in Figure **5**). The phase difference control portion **2** controls a phase difference between the left and right channel signals S_L and S_R so that the relative phase difference be from 140 degrees to 160 degrees (and preferably about 150 degrees) and outputs the phase difference controlled signals S'_L and S'_R for the left and right sound source, respectively.

The signals S'_L and S'_R processed in the above-mentioned manner are respectively supplied to the sound sources S_{SL} and S_{SR}. As a result, with respect to a monophonic signal, the circuit is capable of preventing sound image localization in the head of the listener and creating sound field just as enveloping the listener. Furthermore, with respect to a stereophonic signal, the circuit is capable of performing a processing which does not compromise the sound quality (i.e., a feeling that sound image_of the left and the right surround channels is comfortably localized).

Figure **2** is a block diagram of an audio signal processing circuit **4** which is incorporated into an audio reproduction apparatus, wherein the phase difference control portion **2** includes all pass filters (APFs) **6** and **8**. The apparatus includes an amplifier and speakers both of which are connected

to the output of the audio signal processing circuit **4** (not shown in Figure **2**).

A central channel signal **C**, a front left channel signal $\mathbf{F_L}$, a front right channel signal $\mathbf{F_R}$, a surround left channel signal $\mathbf{S_L}$, a surround right channel signal $\mathbf{S_R}$, and a low frequency channel signal **LFE** are input to the circuit **4**. Among these signals, The central channel signal **C**, the front left channel signal $\mathbf{F_L}$, the front right channel signal $\mathbf{F_R}$, and the low frequency channel signal **LFE** are output without any processing. The surround left channel signal $\mathbf{S_L}$ is processed with the APF **6** so as to be output as the signal $\mathbf{S'_L}$. The surround right channel signal $\mathbf{S_R}$ is processed with the APF **8** so as to be output as the signal $\mathbf{S'_R}$. In this embodiment, the APFs **6** and **8** constitute the phase difference control portion **2**.

An example of the APF **6** is shown in Figure **3A**. The example illustrates secondary APF. A frequency-phase relationship of the APF **6** is shown as a curved line **10** in Figure **4**. In a low frequency region, the phase of the output signal is the same as that of the input signal (i.e., the phase difference between the input and the output signals is zero). The phase of the output signal delays as the frequency increases, and in a high frequency region, the phase of the output signal becomes again the same as that of the input signal (i.e., the

phase difference between the input and the output signals becomes 360 degrees). In other words, the phase difference between the input and the output signals varies in the range of zero to 360 degrees depending upon the frequency. The properties of the APF **6** represented by the curved line **10** may be adapted by selecting resistance **R1** and **R2** and capacitor **C1** and **C2**.

A desired phase difference $\arg(S'_R/S'_L)$ is represented by the following equation:

$$\arg(S'_R/S'_L)=\arg(S'_R/S_R)-\arg(S'_L/S_L)$$

here, the following equations are satisfied:

$$\arg(S'_L/S_L)=\tan^{-1}((-2(f/f1))/(1-(f/f1)2)) + \tan^{-1}((-2(f/f2))/(1-(f/f2)2))$$

$$\arg(S'_R/S_R)=\tan^{-1}((-2(f/f3))/(1-(f/f3)2)) + \tan^{-1}((-2(f/f4))/(1-(f/f4)2))$$

$$f1=1/(2\pi C1*R1)$$

$$f2=1/(2\pi C2*R2)$$

$$f3=1/(2\pi C3*R3)$$

$$f4=1/(2\pi C4*R4).$$

Therefore, the APF **6** having desired properties can be designed based on the above-mentioned equations.

An example of the APF **8** is shown in Figure **3B.** The structure thereof is basically the same as that of the APF **6.** The properties of the APF **8** represented by a curved line **12** of Figure **4** are obtained by selecting resistance **R3** and **R4** and capacitor **C3** and **C4.** By utilizing the above-mentioned APFs **6** and **8,** the phase difference of 140 to 160 degrees can be obtained between the surround left channel signal S'_L and the surround right channel signal S'_R in a frequency region ranging from 200 Hz to 1 kHz. In other words, when the monophonic surround left channel signal S_L and the monophonic surround right channel signal S_R are supplied to the APFs **6** and **8,** the APFs **6** and **8** can control the phase difference between the signals S_L and S_R so that the phase of the signal S'_R relatively progresses or delays 140 to 160 degrees to that of the signal S'_L.

The output signals obtained in the above-mentioned manner are supplied to respective speakers as shown in Figure **5.** More specifically, the central channel signal **C** is supplied to a speaker S_C; the front left channel signal F_L is supplied to a speaker S_{FL}; the front right channel signal F_R is supplied to a speaker S_{FR}; and the low frequency channel signal **LFE** is supplied to a speaker S_{LFE}. Furthermore, the surround left channel signal S'_L is supplied to a speaker S_{SL},

and the surround right channel signal S'_R is supplied to a speaker S_{SR}.

Alternatively, the relative phase difference of 140 to 160 degrees can be obtained by producing a phase difference of 20 to 40 degrees between the channels with APFs and then inversing the phase of one of the channels.

Although the desired phase difference is produced in the frequency region of 200 Hz to 1 kHz according to the above-mentioned embodiment, it is more preferred if the desired phase difference can be obtained in the frequency region of 50 Hz to 4 kHz. The higher order of the APFs widens the frequency band wherein the desired phase difference is obtained.

Although the above-mentioned embodiment has illustrated the case where the surround speakers S_{SL} and S_{SR} are arranged at just the left and the right sides to the listener **50**, the surround speakers S_{SL} and S_{SR} may be arranged in an angular range represented by α of Figure **5**. In Figure **5**, the angle range α of 60 degrees (more specifically, 30 degrees both in front and in rear with respect to the line connecting the surround speakers S_{SL} and S_{SR}) is exemplified. Accordingly, in the present specification, the phrase "substantially at

left and right sides to a listener" is meant to be the above-mentioned angular range α.

Figure **6** shows a surround audio reproduction apparatus creating virtual sound sources with DSP, wherein the phase difference control portion in accordance with the present invention is incorporated. The respective input signals **C**, $\mathbf{F_L}$, $\mathbf{F_R}$, $\mathbf{S_L}$, $\mathbf{S_R}$ and **LFE** are obtained by decoding a digitized data converted from an analog signal with an A/D converter or a digital-bit-stream encoded for surround, with a multi-channel surround decoder (not shown). The respective input signals are supplied to the DSP **22**. The multi-channel surround decoder can either be incorporated into the DSP or separately provided therefrom.

A signal for a left speaker $\mathbf{L_{OUT}}$, a signal for a right speaker $\mathbf{R_{OUT}}$ and a signal for a sub-woofer speaker $\mathbf{SUB_{OUT}}$ are produced by performing processings such as addition, subtraction, filtering, delay and the like with the DSP **22** to the thus-input digital data in accordance with program(s) stored in a memory **26**. The thus-produced signals are converted into analog signals with a D/A converter **24** and are supplied to the speakers $\mathbf{S_{FL}}$, $\mathbf{S_{FR}}$ and $\mathbf{S_{LFE}}$. Installation process of the program(s) into the memory 26 and other processings are carried out by a micro-processor **20**.

In this embodiment, it is presumed that the speakers S_{FL} and S_{FR} and the virtual surround sound sources X_{SL} and X_{SR} are symmetrically arranged with respect to the central axis 40 through the listener as shown in Figure **7**. Since bass (sound having a low frequency) reproduced by the woofer speaker S_{LFE} has a weak directivity and a long wavelength, the woofer speaker S_{LFE} can be arranged at any location.

Figure **8** is a signal-flow diagram illustrating processings carried out by the DSP **22** in accordance with the program(s) stored in the memory **26**. According to this embodiment, as shown in Figure **7**, the virtual central sound source X_C, the virtual surround left sound source X_{SL} and the virtual surround right sound source X_{SR} are created by using only the front left and right speakers S_{FL} and S_{FR} and the low frequency speaker S_{LFE}.

The surround left channel signal S_L and the surround right channel signal S_R are subjected to a sound image localization processing with a surround sound image localization circuit **12** and are supplied to the left and the right speakers S_{FL} and S_{FR} arranged in front of the listener. The surround sound image localization circuit **12** is composed of a so-called shuffler type filter. Therefore, the effect that the surround left channel signal S_L and the surround right channel signal S_R are output respectively from the virtual

surround left sound source $\mathbf{X}_{SL}$ and the virtual surround right sound source $\mathbf{X}_{SR}$ can be obtained.

The central channel signal **C** is equally supplied to the left and the right speakers $\mathbf{S}_{FL}$ and $\mathbf{S}_{FR}$. Therefore, the effect that the central channel signal **C** is output from the virtual central sound source $\mathbf{X}_{C}$ can be obtained.

Delay processing circuits **14L, 14R** and **30** provide a delay time equal to that caused by the surround sound image localization circuit **12**. These delay circuits can compensate the delay between the signals **C**, $\mathbf{F}_{L}$, $\mathbf{F}_{R}$ and **LFE** and the signals $\mathbf{S}_{L}$ and $\mathbf{S}_{R}$.

The surround left channel signal $\mathbf{S}_{L}$ and the surround right channel signal $\mathbf{S}_{R}$ are subjected to a phase difference control processing with the phase difference control portion **2** in the above-mentioned manner before being supplied to the surround sound image localization circuit **12**. Therefore, a relative phase difference of 140 to 160 degrees has already been produced between the surround left channel signal $\mathbf{S}_{L}$ and the surround right channel signal $\mathbf{S}_{R}$.

In this embodiment, a secondary IIR filter as shown in Figure **9** is used as the APFs **6** and **8** constituting the phase difference control portion **2**.

Since the phase difference control processing is performed with the phase difference control portion **2**, the surround left channel signal $\mathbf{S_L}$ output from the virtual surround left sound source $\mathbf{X_{SL}}$ and the surround right channel signal $\mathbf{S_R}$ output from the virtual surround right sound source $\mathbf{X_{SR}}$ may be prevented from being localized in the head of the listener **50**.

Figure **10** is a signal-flow diagram according to another embodiment of the present invention. According to this embodiment, the front left channel signal $\mathbf{F_L}$ and the front right channel signal $\mathbf{F_R}$ are respectively added to the surround left channel signal $\mathbf{S_L}$ and the surround right channel signal $\mathbf{S_R}$ which have already been subjected to the phase difference control processing. As a result, as shown in Figure **11**, the front left channel signal $\mathbf{F_L}$ is localized at the position of the virtual sound source $\mathbf{X_{FL}}$ located between the positions of the left speaker $\mathbf{S_{FL}}$ and the virtual surround left sound source $\mathbf{X_{SL}}$. Likewise, the front right channel signal $\mathbf{F_R}$ is localized at the position of the virtual sound source $\mathbf{X_{FR}}$ located between the positions of the right speaker $\mathbf{S_{FR}}$ and the virtual surround right sound source $\mathbf{X_{SR}}$. Accordingly, sound field created by the front left channel signal $\mathbf{F_L}$ and the front right channel signal $\mathbf{F_R}$ can be widen.

In the above embodiments, an analog circuit can be used in place of the described digital circuit and a digital circuit can be used in place of the described analog circuit.

Figure **12** is a schematic view of a shuffler type cross-talk cancel filter **130** according to an embodiment of the present invention. A left channel signal is supplied to a left channel input terminal $\mathbf{L_{IN}}$ and a right channel signal is supplied to a right channel input terminal $\mathbf{R_{IN}}$. The left and the right channel signals are added up with an adder **122** and the added signal is supplied to a first filter **120a**. The right channel signal is subtracted from the left channel signal with a subtracter **124** and the subtracted signal is supplied to a second filter **120b**. Transfer functions H_{SUM} and H_{DIF} of the first and the second filters **120a** and **120b** are represented by the following equations, respectively:

$$H_{SUM}=ha/2(ha+hb)$$

$$H_{DIF}=ha/2(ha-hb)$$

An adder **126** adds the outputs of the first and the second filters **120a** and **120b** and outputs a signal for a speaker **104L**. A subtracter **128** subtracts the outputs of the second filter **120b** from the output of the first filter **120a** and outputs a signal for a speaker **104R**.

According to this embodiment, the first and the second filters **120a** and **120b** are FIR filters and the cross-talk cancel filter **130** is composed of DSP. Figure **13** is a block diagram illustrating a hardware structure of the audio reproduction apparatus using DSP **140**. A left and a right channel signals **L** and **R** are supplied as digital data to the DSP **140**. A signal for a left speaker $\mathbf{L_{OUT}}$ and a signal for a right speaker $\mathbf{R_{OUT}}$ are produced by performing processings such as addition, subtraction, filtering, delay and the like with the DSP **140** to the thus-input digital data in accordance with program(s) stored in a memory **146**. The thus-produced signals are converted into analog signals with a D/A converter **142** and are supplied to the speakers **104L** and **104R**. Installation process of the program(s) into the memory 26 and other processings are carried out by a micro-processor **120**.

Figure **14** is a signal-flow diagram illustrating processings carried out by the DSP **140** in accordance with the program(s) stored in the memory **146**. According to this embodiment, the first and the second filters **120a** and **120b** are FIR filters. In Figure **14**, **DS1** to **DS31** and **DD1** to **DD95** denote delay means. The delay means perform delay processing in an amount of one sampling data. In this embodiment, the sample frequency is set to be 48 kHz. **KS0** to **KS31** and **KD0** to K**D95** denote coefficient processing means. In this embodiment, the tap number (i.e., the number of the

coefficient processings) of the first filter **120a** is set to be 32 and the tap number of the second filter **120b** is set to be 96. In the case of FIR filter, the larger tap number produces the higher accuracy in a low frequency region. Accordingly, in the example of Figure **14**, the accuracy of the second filter **120b** is higher than that of the first filter **120a** in a low frequency region.

Figure **15** shows a frequency response H_{SUM} of the first filter **120a** and a frequency response H_{DIF} of the second filter **120b** wherein the first and the second filters have 32 taps. Figure **15** also shows a cross-talk cancel response Zt1 and a cross-talk cancel error Zt2 when a cross-talk cancel filter wherein the first and the second filters are incorporated is used. Here, the error is meant to be a remained response (i.e., a response that had not been sufficiently canceled). Therefore, regarding the cross-talk cancel filter, the better filter produces the smaller error. In this embodiment, an angle β defined by the speaker **104L** (or **104R**) and the listener **102** as shown in Figure **12** is set to be 10 degrees. As shown in Figure **15**, when the tap number of the first and the second filters **120a** and **120b** is 32,the accuracy is low and a large cross-talk cancel error is caused.

Figure **16** shows a frequency response H_{SUM} of the first filter **120a** and a frequency response H_{DIF} of the second filter

120b wherein the first and the second filters have 64 taps. Figure **16** also shows a cross-talk cancel response Zt1 and a cross-talk cancel error Zt2 when a cross-talk cancel filter wherein the first and the second filters are incorporated is used. Figure **16** shows that, although the cross-talk cancel properties are improved compared to the case of 32 taps shown in Figure **15**, the cross-talk cancel error is still large.

Figure **17** shows a case where the first and the second filters **120a** and **120b** have 96 taps. Figure **17** shows that the cross-talk cancel error is small. However, in this case, the problem that an arithmetical load to DSP **140** is large arises.

According to this embodiment, the tap number of the first filter **120a** is set to be smaller than that of the second filter **120b** in view of the fact that a frequency response required for the first filter **120a** is low level and flat especially in a low frequency region. In other words, the accuracy of the first filter **120a** is set to be low in a low frequency region and the accuracy of the second filter **120b** is set to be higher instead. More specifically, the tap number of the first filter **120a** is set to be 32 and the tap number of the second filter **120b** is set to be 96. Frequency response H_{SUM} and H_{DIF}, a cross-talk cancel response zt1 and a cross-talk cancel error zt2 in this case are shown in Figure **18**.

As is apparent from Figure **18**, the error in this case is as small as that in the case where the tap numbers of the first and the second filters **120a** and **120b** are both 96. According to this embodiment, a shuffler type cross-talk cancel filter having high accuracy can be obtained while keeping low a total tap number thereof.

Figure **19** is a signal-flow diagram according to another embodiment of the present invention. FIR filters are also employed in this embodiment. Furthermore, the tap number of the second filter **120b** is set to be larger than that of the first filter **120a**. More specifically, the tap number of the second filter **120b** is set to correspond to 128 and the tap number of the first filter **120a** is set to be 32. In addition, a filter bank is employed for the second filter **120b** according to this embodiment. As a result, down-sampling is performed with respect to the signal supplied to the second filter **120b** and then the signal is processed with the FIR filters. In figure **19**, **H** denotes a high-pass filter, **G** denotes a low-pass filter, the arrow ↓ denotes down-sampling by 2 and the arrow ↑ denotes up-sampling by 2. Delay means **205**, **206** and **208** perform delay processing which compensates a time required for the processing performed by the filter bank. The delay means **205** performs delay processing in an amount of three sampling data, the delay means **206** performs delay processing in an amount of one sampling data, and the delay

means **208** performs delay processing in an amount of seven sampling data.

According to this embodiment employing the filter bank, a cross-talk cancel filter having a high ability of 128 taps can be obtained while the total tap number of the FIR filters **201, 202, 203** and **204** is kept 68 taps. In other words, a processing margin can be increased by performing down-sampling. As a result, the accuracy in a low frequency component can be improved. Although a so-called octave dividing filter bank has been exemplified in this embodiment, a so-called equal dividing filter bank may also be employed. According to the octave dividing filter bank, a frequency component is divided in a geometrical ratio preferentially in a lower frequency side. In contrast, according to the equal dividing filter bank, a frequency component is equally divided with respect to an overall frequency region.

Figure **20** shows a cross-talk cancel error ZT2 in the case where the tap number of the first filter **120a** is 32 and the tap number of the second filter **120b** is 128 and where a filter bank is not employed. Figure **21** shows a cross-talk cancel error ZT2 when the cross-talk cancel filter shown in Figure **19** is used. As is apparent from the comparison between Figures **20** and **21**, the circuit of Figure **19** which employs a filter

bank has the ability as good as that of the circuit having actually 128 taps.

Figure **22** is a signal-flow diagram according to still another embodiment of the present invention. According to this embodiment, the first filter **120a** is FIR filter having 32 taps and the second filter **120b** is composed of a parallel connection of FIR filter **210** having 32 taps and secondary IIR filter **212**. The outputs of the FIR filter **210** and the secondary IIR filter **212** are added up with an adder **214**.

According to this embodiment, an accuracy with respect to a low frequency component can be improved by utilizing the secondary IIR filter **212** while the tap number of the FIR filter **210** in the second filter is kept 32 taps. Since the secondary IIR filter produces a higher accuracy in a low frequency region, the cross-talk cancel filter according to this embodiment produces an accuracy as high as the filter of Figure **12** wherein both of the first and the second filters are FIR filters, while the tap number of the filter according to this embodiment is smaller than that of the filter of Figure **12**. Although the secondary IIR filter has been exemplified in this embodiment, IIR filter of the first order or the higher order may also be employed. The IIR filter of the higher order can be composed of either series connection or parallel connection.

Figure **23** shows a frequency response H_{SUM} of the first filter **120a** and a frequency response H_{DIF} of the second filter **120b** in the circuit (i.e., the cross-talk cancel filter) of Figure **22**. Figure **23** also shows a cross-talk cancel response Zt1 and a cross-talk cancel error Zt2 of the circuit of Figure **22**. As is apparent from Figure **23**, accuracy substantially as high as that of the case shown in Figure **18** is obtained.

According to the embodiment shown in Figure **22**, the second filter **120b**, which is composed of parallel connection of the FIR filter and the secondary IIR filter, is exemplified. However, as shown in Figure **24**, one of intermediate taps of the FIR filter can be connected to the input of the secondary IIR filter. The end tap (i.e., the tap of the number m-1 in Figure **24**) may also be connected to the input of the secondary IIR filter. As a result, properties of the second filter **120b** can be easily varied depending upon the desired properties.

Hereinafter, a design method of the filter shown in Figure **24** will be described with reference to Figures **25** to **28**. Figure **25** shows an impulse response required for the second filter **120b**. Based on the required impulse response, an impulse response of the secondary IIR filter is decided. Initially, the impulse response is decided by preferentially

approximating it to the latter part of the required impulse response (which corresponds to a low frequency region), as shown in Figure **26**. In the example of Figure **26**, the impulse response of the secondary IIR filter having the property approximate to that of the required impulse response after the sample of the number k is obtained. It is noted that; with respect to the sample of the number k to the sample of the number m, the impulse response of the secondary IIR filter is largely deviated from the required impulse response.

Next, the impulse response of the FIR filter is obtained with respect to the sample of the number zero to the sample of the number m. As described above and as shown in Figure **27**, the impulse response of the secondary IIR filter is largely deviated from the required impulse response with respect to the sample of the number k to the sample of the number m. In consideration of such a deviation, the impulse response of the FIR filter as shown in Figure **28** is obtained with respect to the sample of the number zero to the sample of the number m.

As described above, the second filter **120b** as shown in Figure **24** can be obtained. The intermediate tap connected to the input of the secondary IIR filter is the tap corresponding to the first sample from which the approximation is conducted (i.e., the sample of the number

k in the above-mentioned example). As described above, a filter having a desired impulse response can be easily obtained.

In the above embodiments, the tap number has been described only for being exemplified. Furthermore, the cross-talk cancel filter has been described in the above embodiments, however, the present invention is applicable to a sound image localization filter.

In the above embodiments, FIR filter is used for the first filter **120a**. However, the first filter **120a** may also be composed of a parallel connection of FIR filter and IIR filter (as shown in Figures **22** and **24**). Alternatively, the first filter **120a** may employ a filter bank. Even in this case, when the second filter **120b** having a higher accuracy than that of the first filter **120a** is employed, a cross-talk cancel filter having a high accuracy can be obtained while keeping simple an overall structure of the filter.

In the above embodiments, although DSP is used in the cross-talk cancel filter, an analog filter may be entirely or partially substituted for the DSP.

Various other modifications will be apparent to and can be readily made by those skilled in the art without departing

from the scope and spirit of this invention. Accordingly, it is not intended that the scope of the claims appended hereto be limited to the description as set forth herein, but rather that the claims be broadly construed.

WHAT IS CLAIMED IS:

1. An audio signal processing circuit for an audio reproduction apparatus at least having sound source located substantially at left and right sides to a listener, comprising:

a phase difference control portion which receives a left channel signal for the left sound source and a right channel signal for the right sound source, controls a phase difference between the left and right channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference controlled left and right channel signals for the left and right sound source, respectively.

2. An audio signal processing circuit according to claim 1, wherein the phase difference control portion produces the relative phase difference of 140 degrees to 160 degrees in a frequency region ranging from 200 Hz to 1 kHz.

3. A surround audio reproduction apparatus having a left and a right channels in front of a listener and a left and a right surround channels at left and right sides with respect to the listener, comprising:

a phase difference control portion which receives a left surround channel signal and a right surround channel signal,

controls a phase difference between the left and the right surround channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference controlled surround left and right channel signals for a left and a right surround sound source, respectively.

4. A surround audio signal processing circuit according to claim 3, wherein the left and the right surround sound sources are a virtual sound source produced by a sound image localization processing.

5. A surround audio signal processing circuit according to claim 3, wherein the phase difference control portion produces the relative phase difference of 140 degrees to 160 degrees in a frequency region ranging from 200 Hz to 1 kHz.

6. An audio reproduction method at least utilizing sound source located substantially at left and right sides to a listener, comprising the steps of:

controlling a phase difference between a left channel signal for the left sound source and a right channel signal for the right sound source so as to produce a relative phase difference in the range of 140 degrees to 160 degrees; and

outputting the phase difference controlled left and right channel signals for the left and right sound source, respectively.

7. A shuffler type audio signal processing circuit, comprising:

a first filter for producing a sum signal of a left channel signal and a right channel signal; and

a second filter for producing a differential signal of the left channel signal and the right channel signal;

wherein an accuracy of the second filter is higher than that of the first filter in a low frequency region.

8. A shuffler type audio signal processing circuit, comprising:

a first filter for producing a sum signal of a left channel signal and a right channel signal; and

a second filter for producing a differential signal of the left channel signal and the right channel signal;

wherein the first filter and the second filter are FIR filter, and the tap number of the second filter is larger than that of the first filter.

9. A shuffler type audio signal processing circuit according to claim 7, wherein the second filter is composed of a filter bank.

10. A shuffler type audio signal processing circuit according to claim 9, wherein the filter bank performs down-sampling by the larger number for the lower frequency component.

11. A shuffler type audio signal processing circuit, comprising:

a first filter for producing a sum signal of a left channel signal and a right channel signal; and

a second filter for producing a differential signal of the left channel signal and the right channel signal;

wherein the first filter is FIR filter, and the second filter is composed of a parallel connection of FIR filter and secondary IIR filter.

12. A shuffler type audio signal processing circuit according to claim 11, wherein the second filter comprises:

FIR filter, and

secondary IIR filter connected in parallel to the FIR filter at one of the intermediate taps or the end tap thereof.

13. An audio signal processing circuit according to claim 7, wherein the circuit is used as a cross-talk cancel filter.

14. An audio signal processing circuit according to claim 7, wherein the circuit is used as a sound image localization processing filter.

15. A filter comprising:

FIR filter having a plurality of taps,

IIR filter whose input is connected to one of the intermediate taps or the end tap of the FIR filter, and

an adding means which adds outputs of the FIR filter and the IIR filter.

16. A shuffler type audio signal processing method, comprising the steps of:

performing a first filtering process for a sum signal of a left channel signal and a right channel signal; and

performing a second filtering process for a differential signal of the left channel signal and the right channel signal

wherein an accuracy of the second filtering process is higher than that of the first filtering process.

ABSTRACT OF THE DISCLOSURE

An audio signal processing circuit for an audio reproduction apparatus at least having sound source located substantially at left and right sides to a listener, is provided. The audio signal processing circuit includes a phase difference control portion. The phase difference control portion receives a left channel signal for the left sound source and a right channel signal for the right sound source, controls a phase difference between the left and right channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference controlled left and right channel signals for the left and right sound source, respectively.

FIG.1

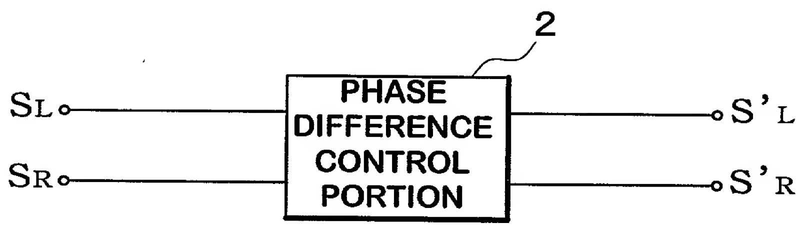

2
SL
SR
PHASE DIFFERENCE CONTROL PORTION
S'L
S'R

FIG.2

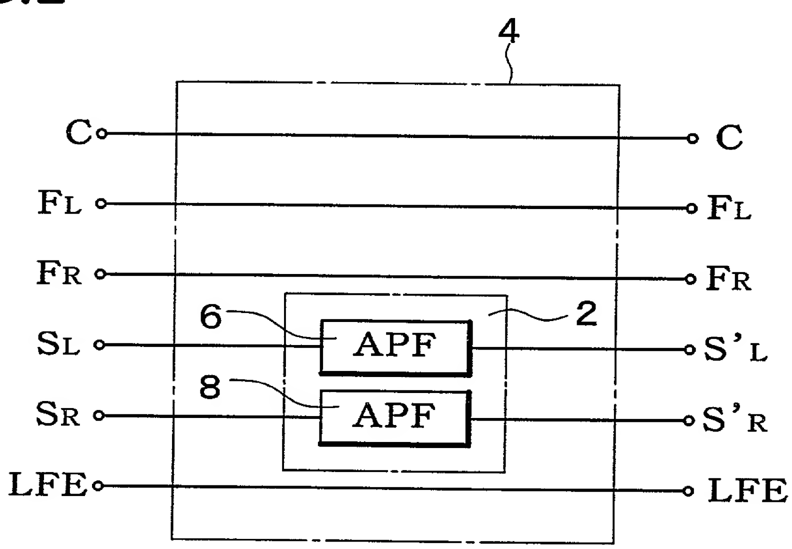

4
C
FL
FR
SL
SR
LFE
6
8
2
APF
APF
C
FL
FR
S'L
S'R
LFE

FIG.3A

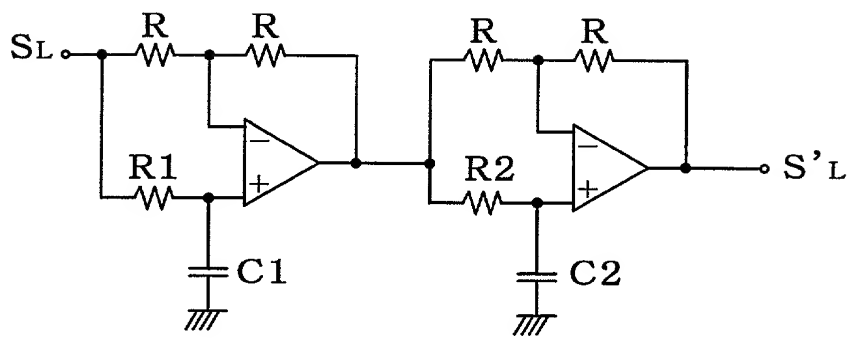


FIG.3B

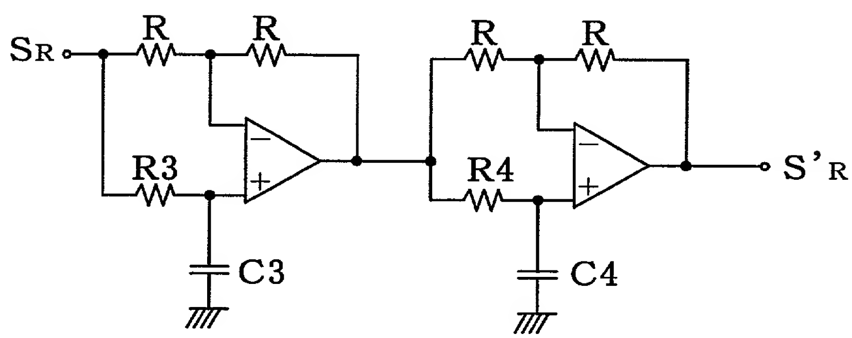


FIG.4

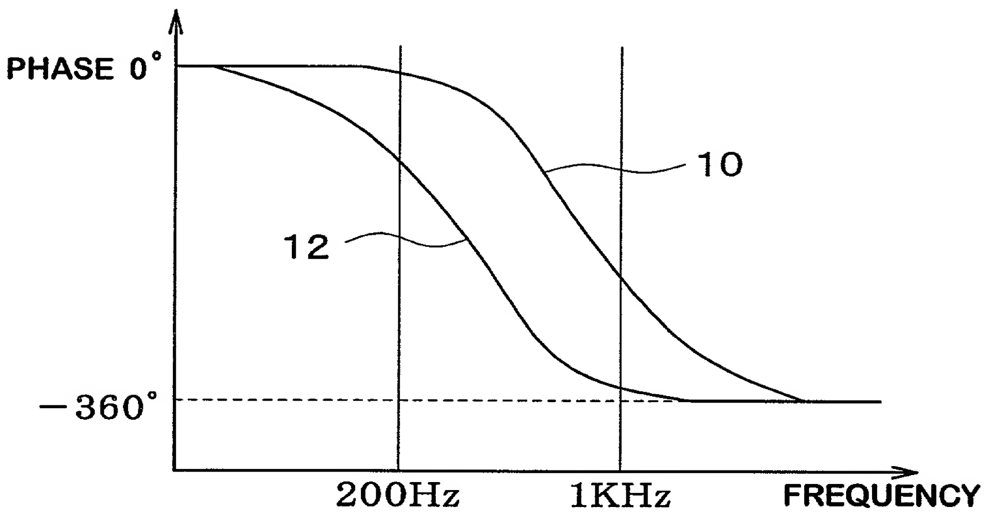


FIG.5

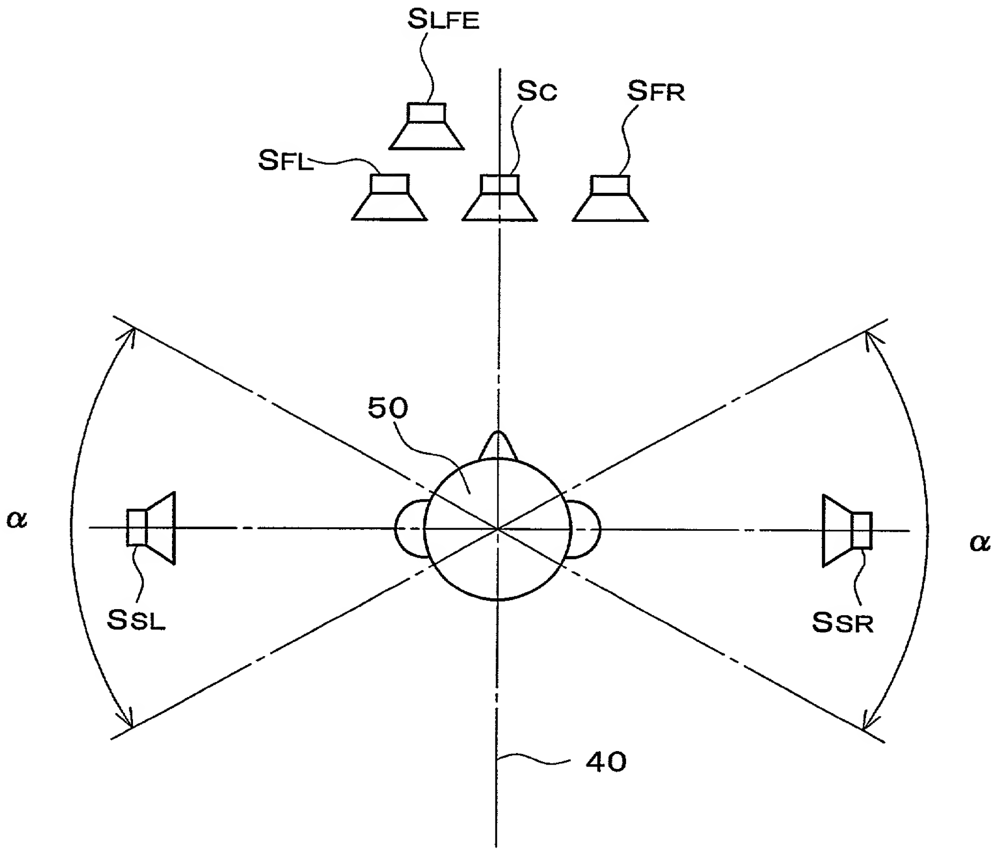
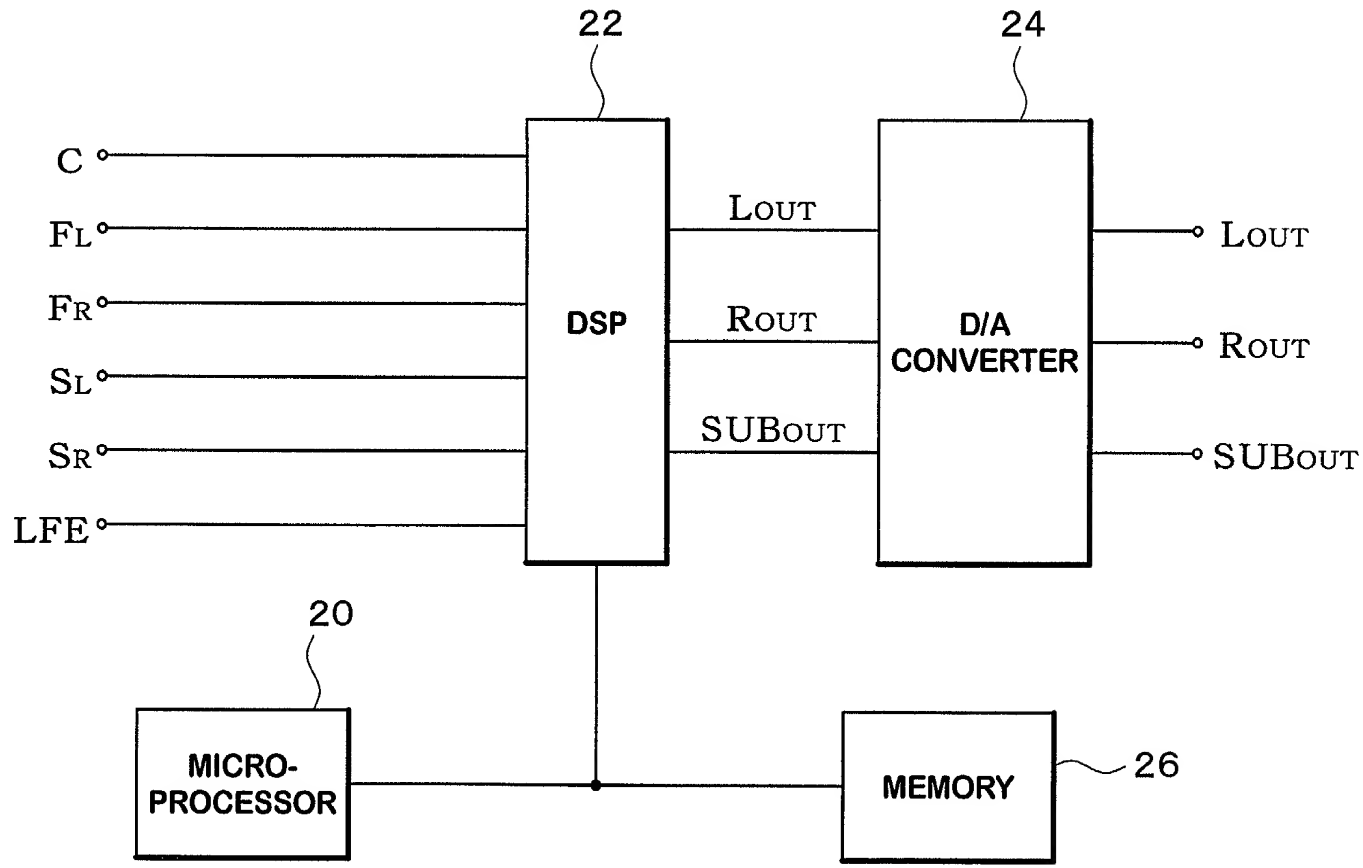


FIG.6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24
C
F_L
F_R
S_L
S_R
LFE
DSP
L_{OUT}
R_{OUT}
SUB_{OUT}
D/A
CONVERTER
L_{OUT}
R_{OUT}
SUB_{OUT}
20
MICRO-
PROCESSOR
MEMORY
26

FIG.7

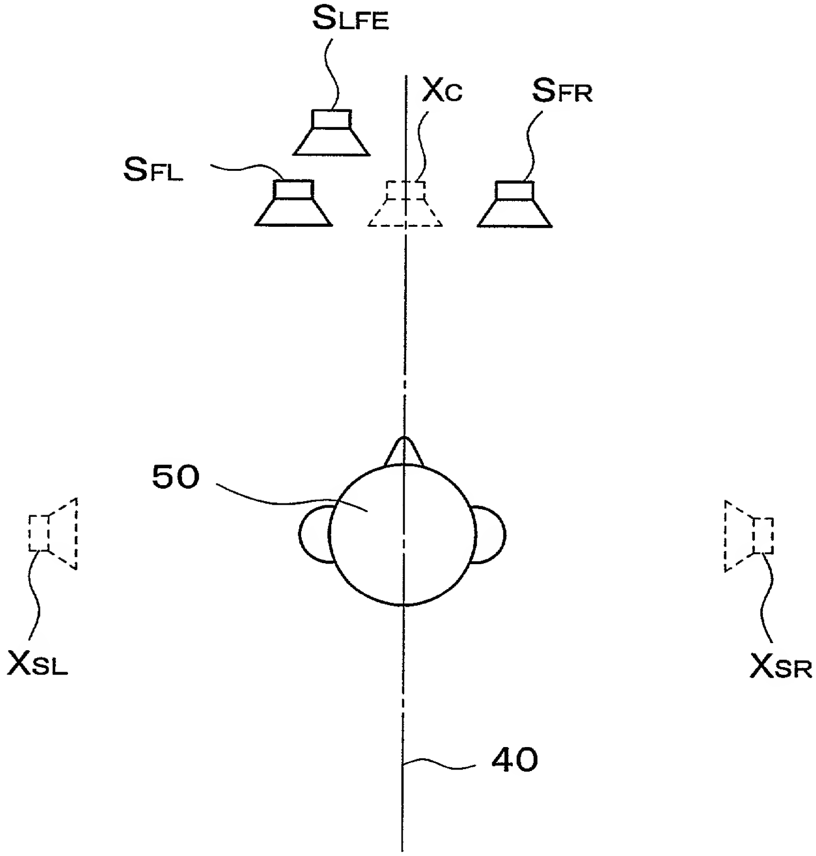
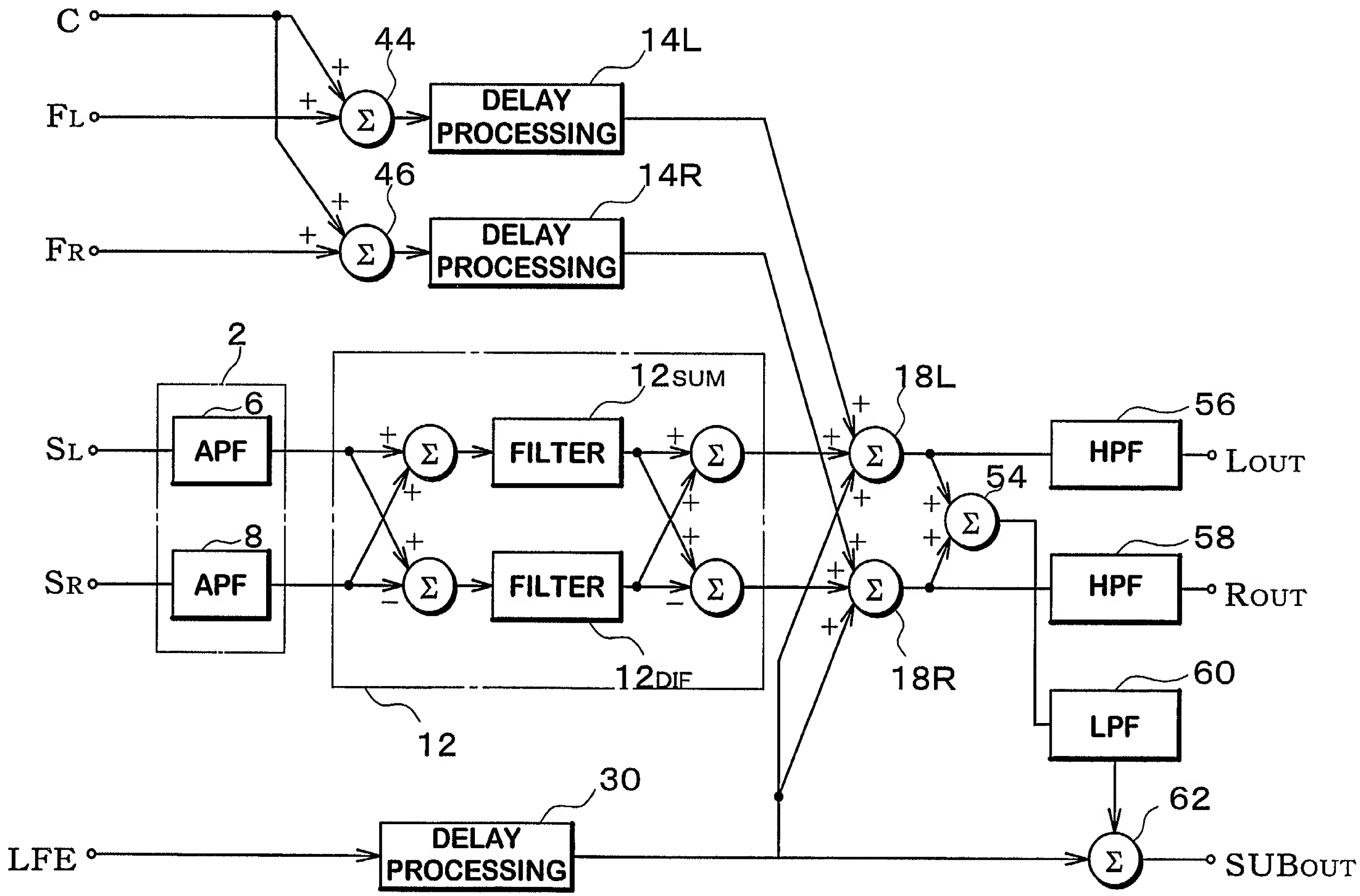
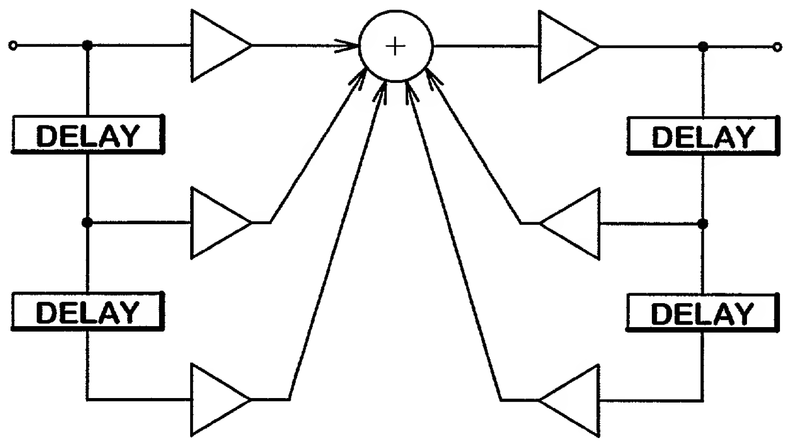


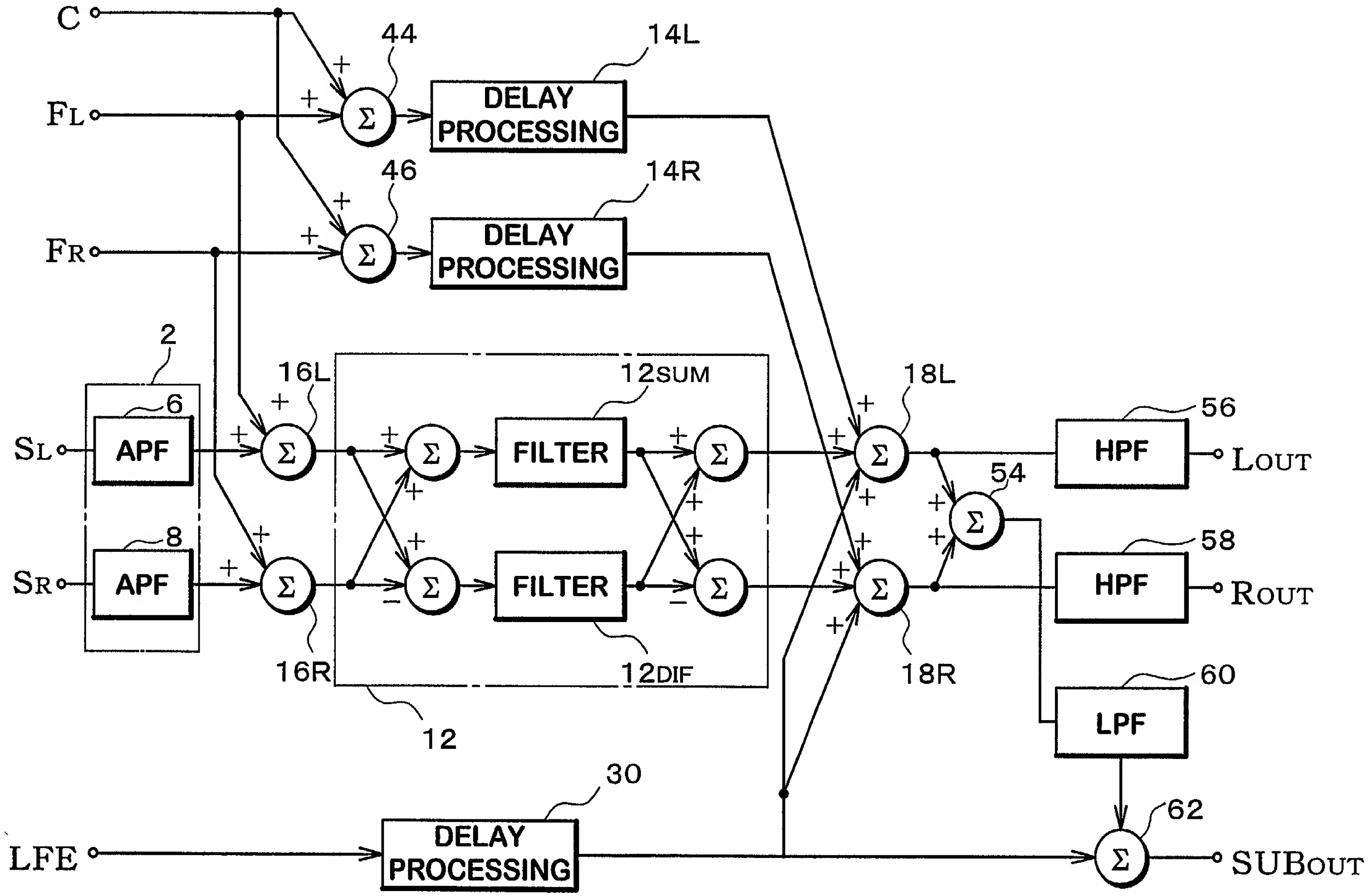
FIG.8



C
FL
FR
SL
SR
LFE
44
46
14L
14R
DELAY PROCESSING
DELAY PROCESSING
2
6
8
APF
APF
12
12SUM
12DIF
FILTER
FILTER
18L
18R
54
56
58
60
62
HPF
HPF
LPF
30
DELAY PROCESSING
LOUT
ROUT
SUBOUT

FIG.9




C
FL
FR
SL
SR
LFE
44
46
14L
14R
DELAY PROCESSING
DELAY PROCESSING
2
6
8
APF
APF
16L
16R
12SUM
12DIF
FILTER
FILTER
12
30
DELAY PROCESSING
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18R
54
56
58
60
62
HPF
HPF
LPF
LOUT
ROUT
SUBOUT

FIG.10

FIG.11

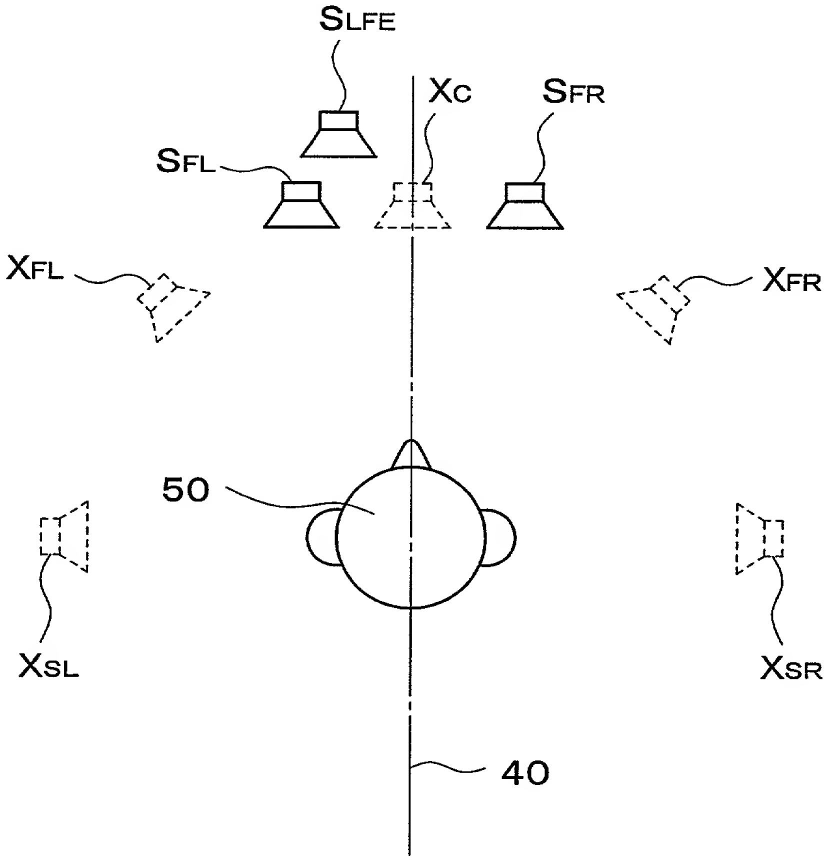


FIG.12

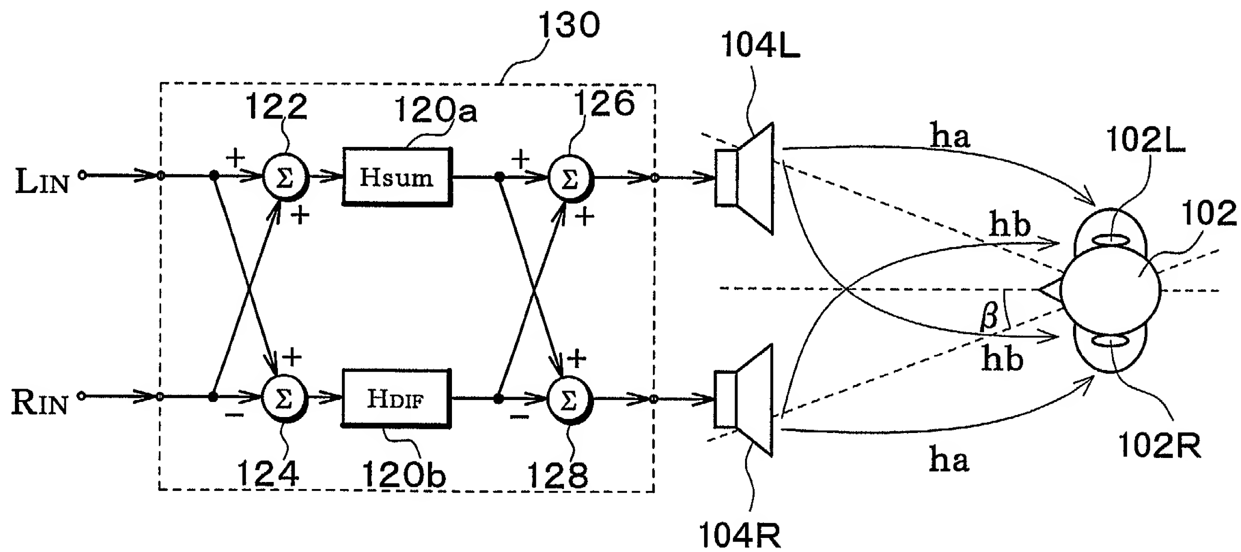

130
104L
122
120a
126
ha
102L
102
LIN
Hsum
hb
β
hb
RIN
HDIF
102R
124
120b
128
ha
104R

FIG.13

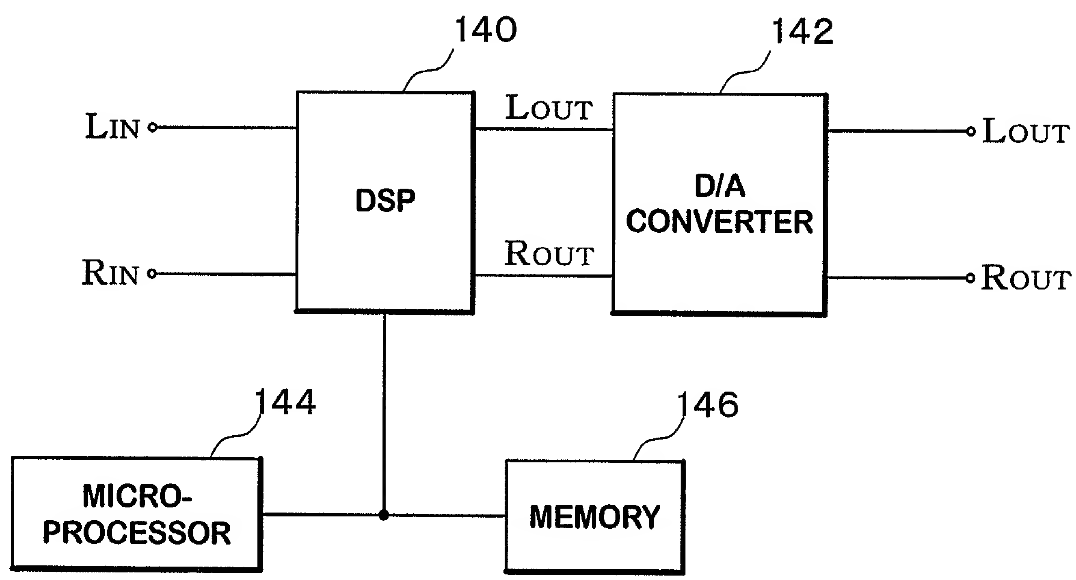

140
142
LIN
RIN
DSP
LOUT
ROUT
D/A
CONVERTER
LOUT
ROUT
144
146
MICRO-
PROCESSOR
MEMORY

FIG.14

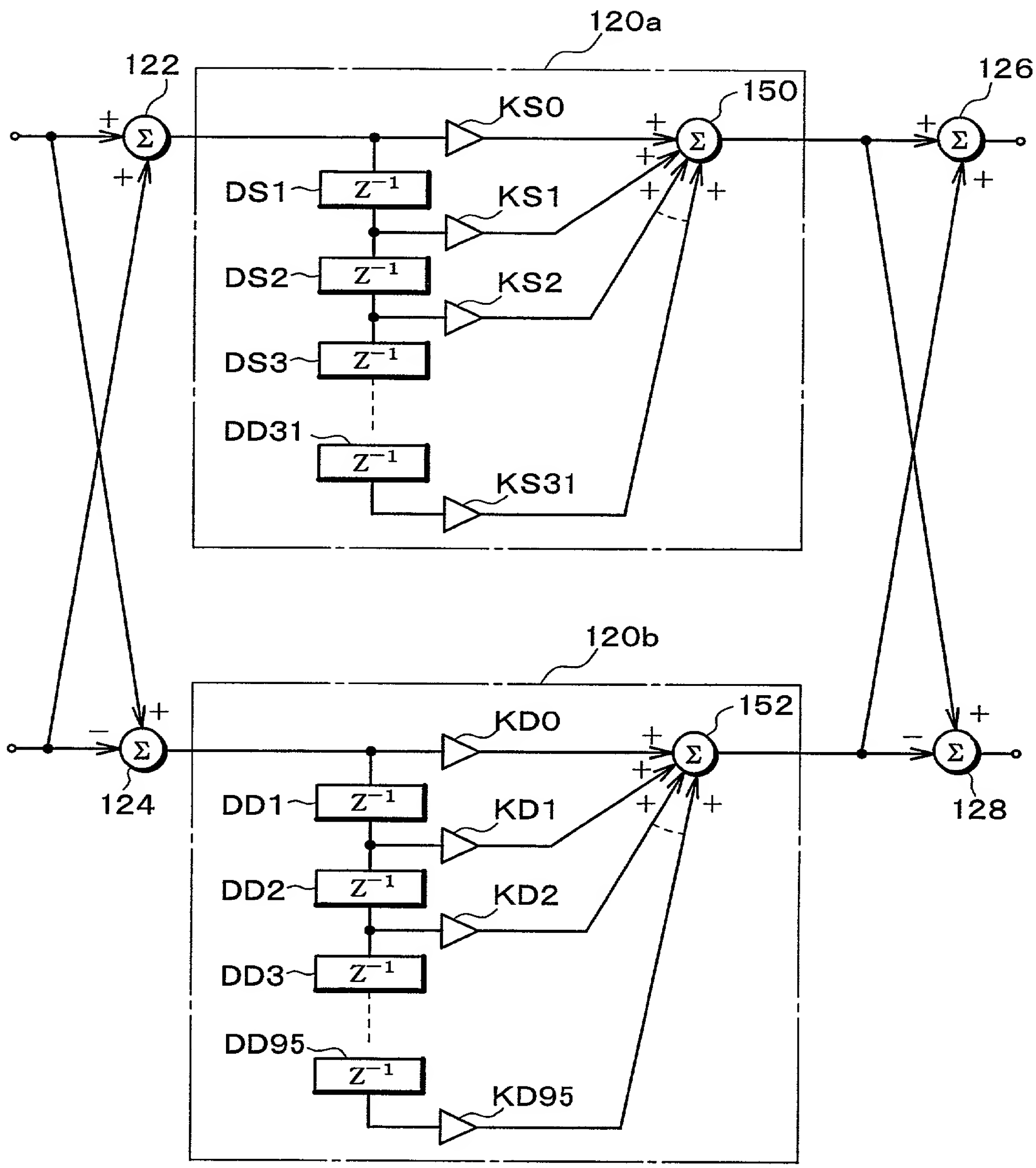

120a
122
150
126
KS0
DS1
KS1
DS2
KS2
DS3
DD31
KS31
120b
152
KD0
124
128
DD1
KD1
DD2
KD2
DD3
DD95
KD95
Z^{-1}
Σ

FIG.15

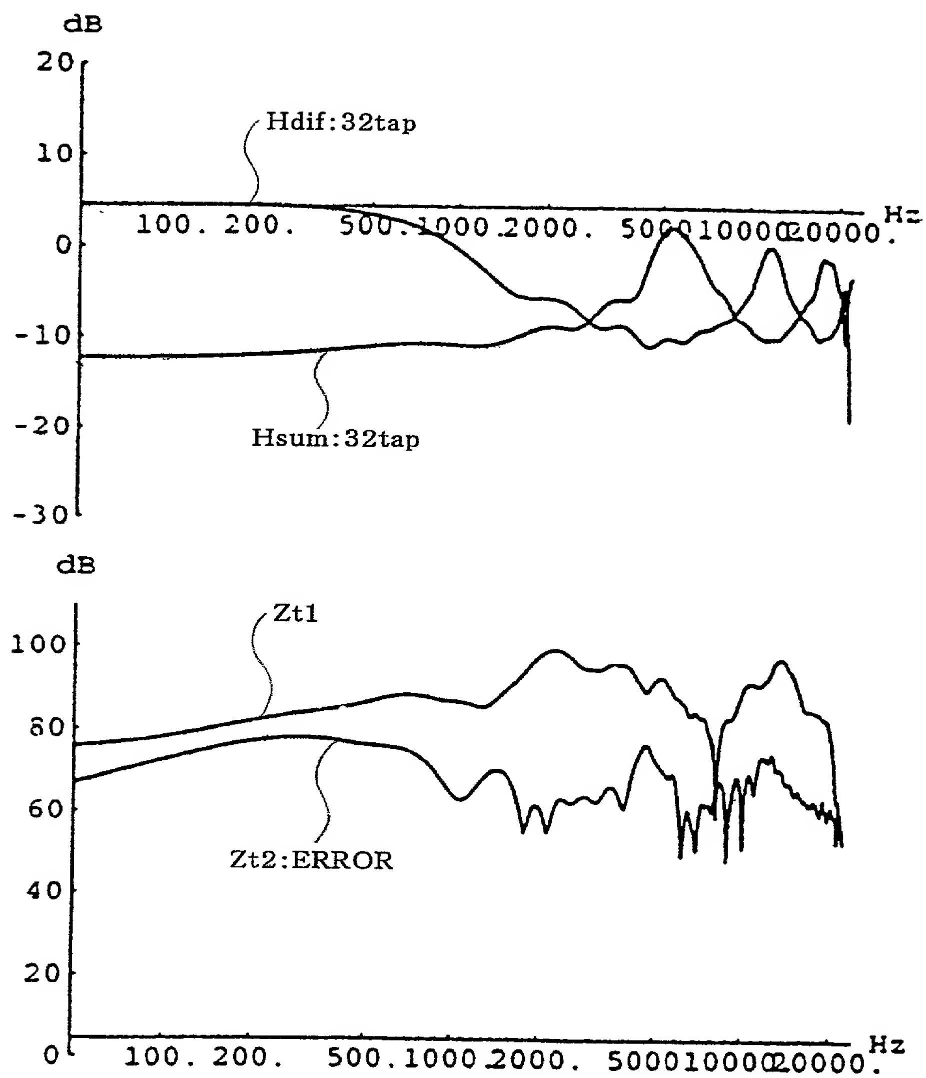
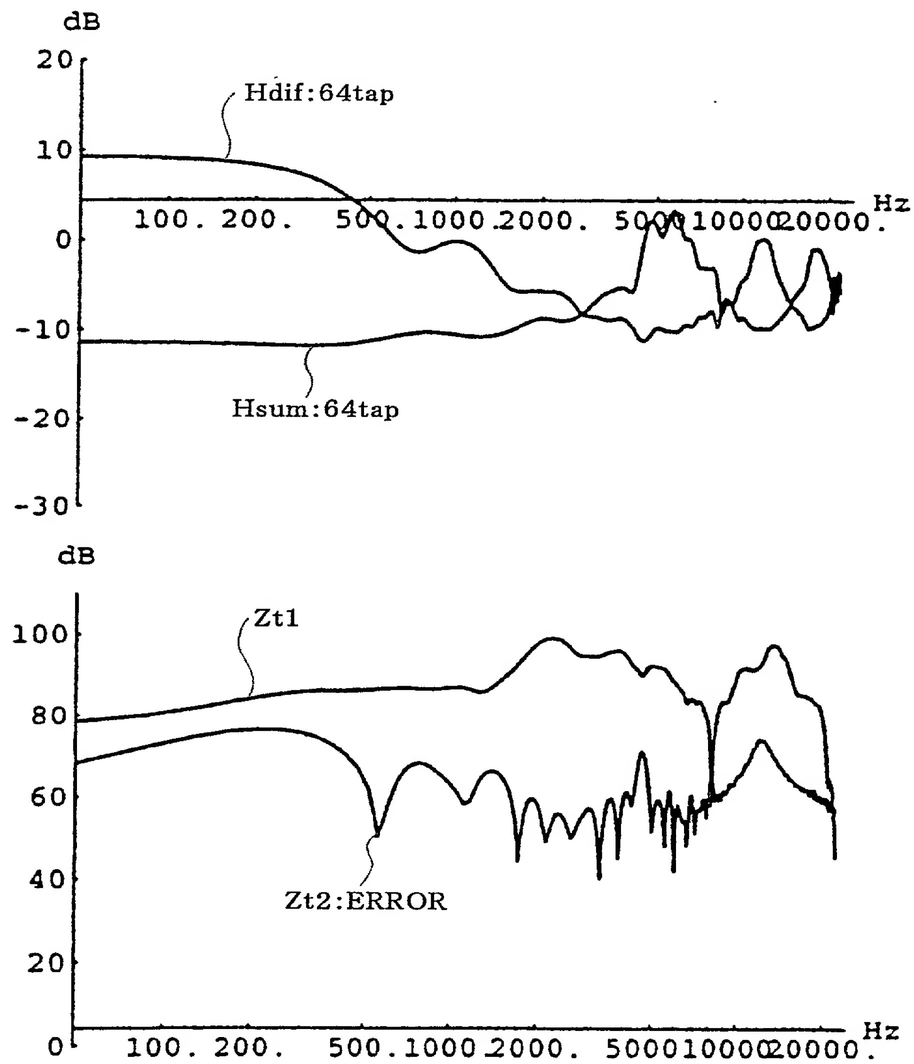

dB
20
10
0
-10
-20
-30
Hdif:32tap
Hsum:32tap
100. 200. 500. 1000. 2000. 5000 10000 20000.
Hz
dB
100
80
60
40
20
0
Zt1
Zt2:ERROR
100. 200. 500. 1000. 2000. 5000 10000 20000.
Hz

FIG.16



dB
20
10
0
-10
-20
-30
Hdif:64tap
Hsum:64tap
100. 200. 500.1000.2000. 50001000020000.
Hz
dB
100
80
60
40
20
0
Zt1
Zt2:ERROR
100. 200. 500.1000 2000. 50001000020000.
Hz

FIG.17

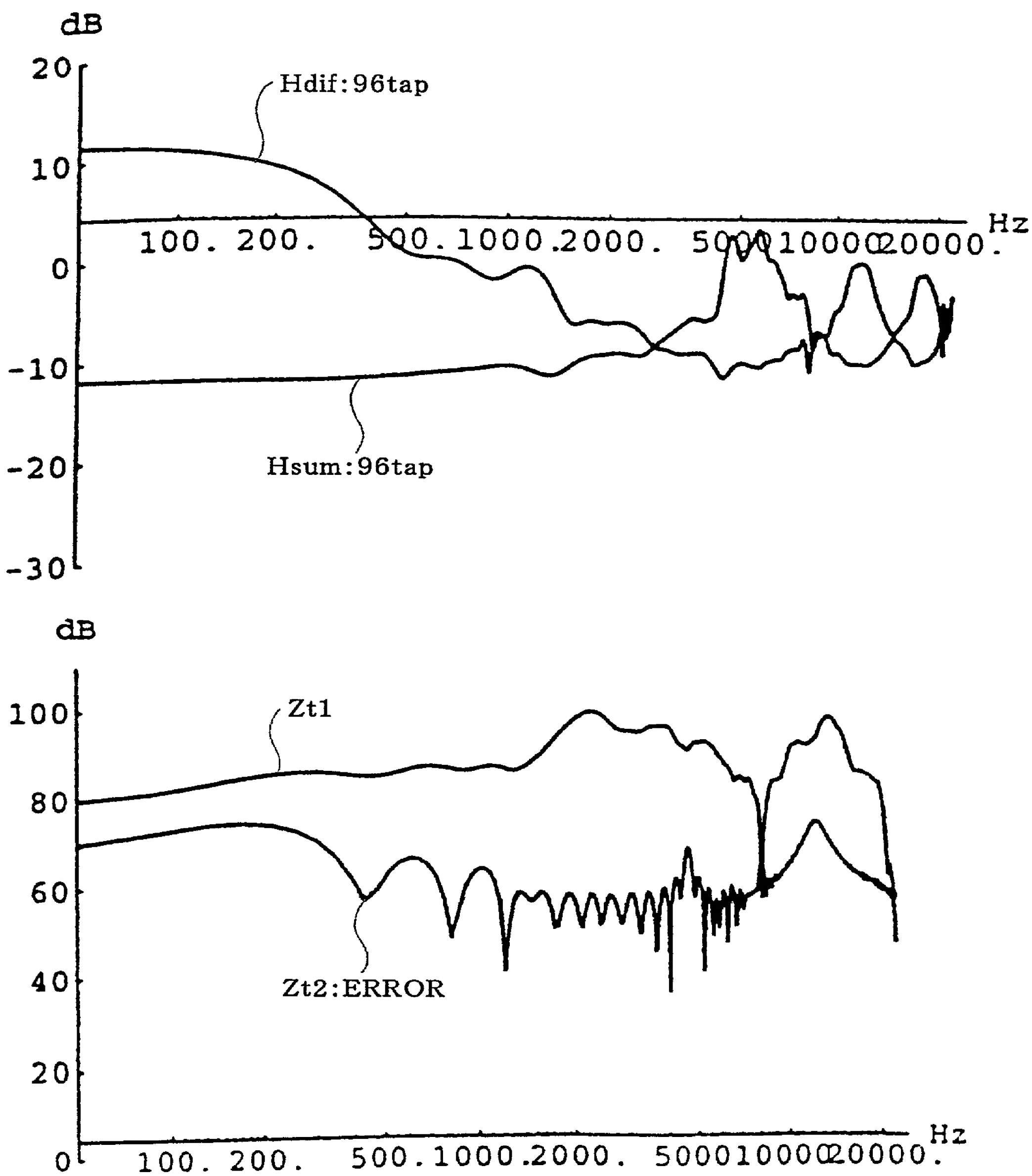

dB
20
10
0
-10
-20
-30
Hdif:96tap
Hsum:96tap
100. 200. 500. 1000. 2000. 5000 10000 20000.
Hz
dB
100
80
60
40
20
0
Zt1
Zt2:ERROR
100. 200. 500. 1000. 2000. 5000 10000 20000.
Hz

FIG.18

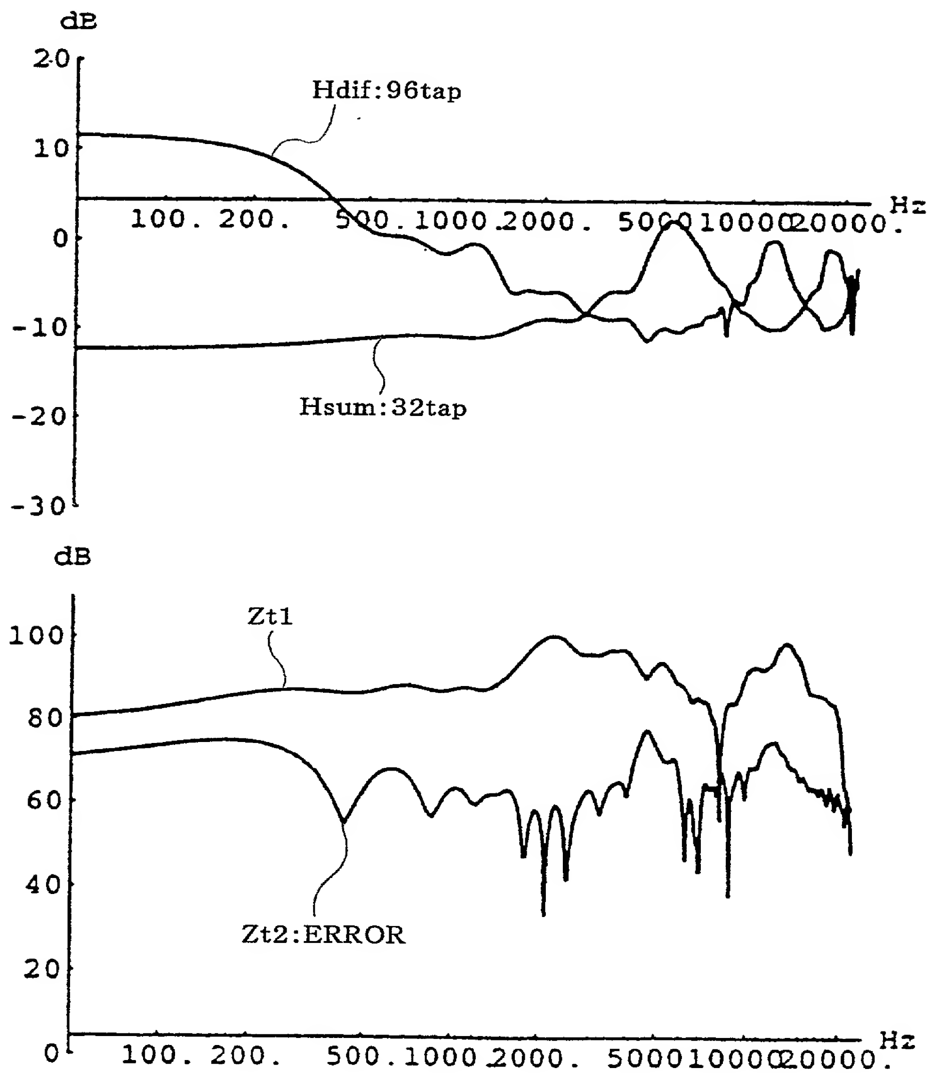
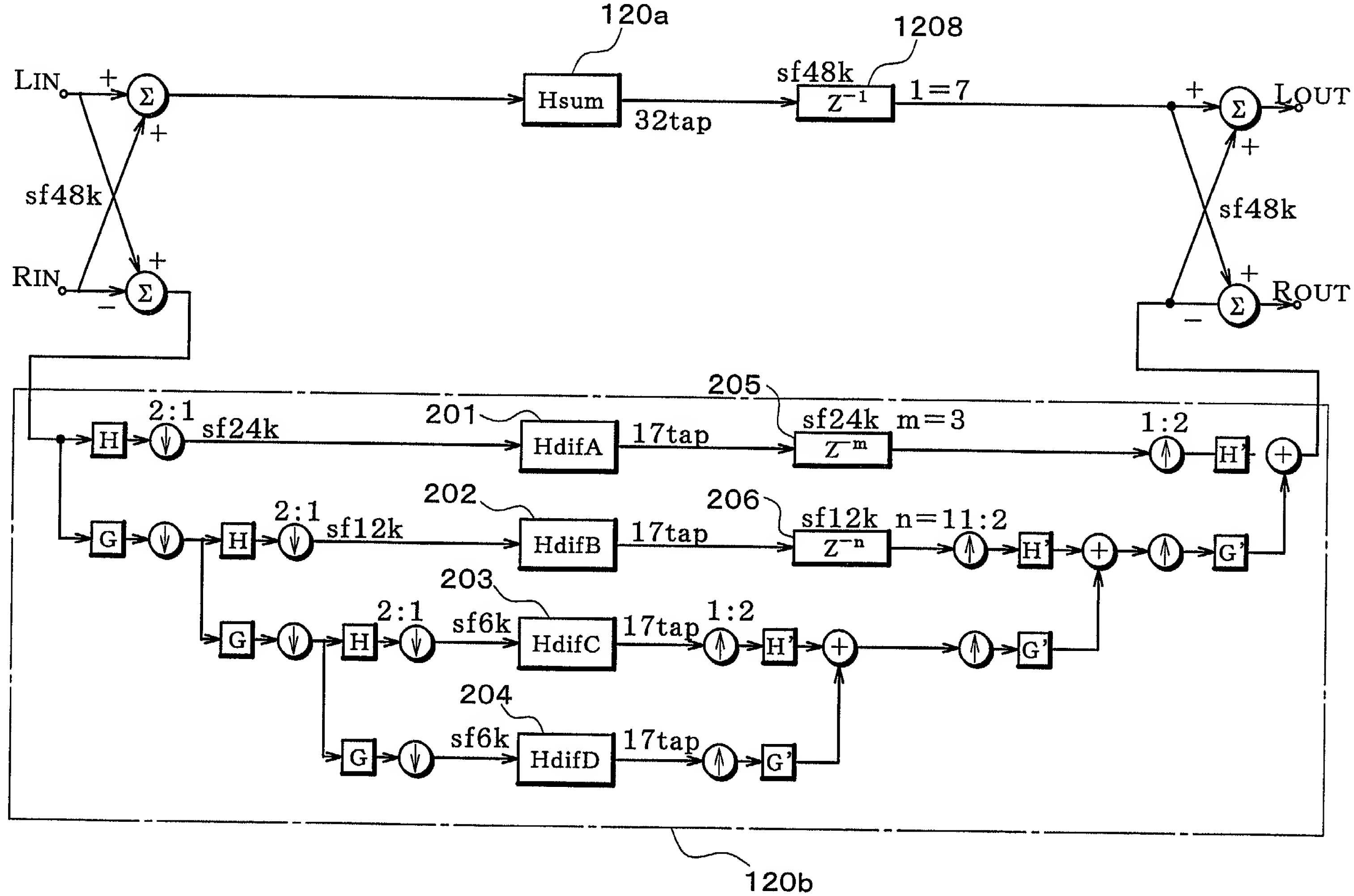
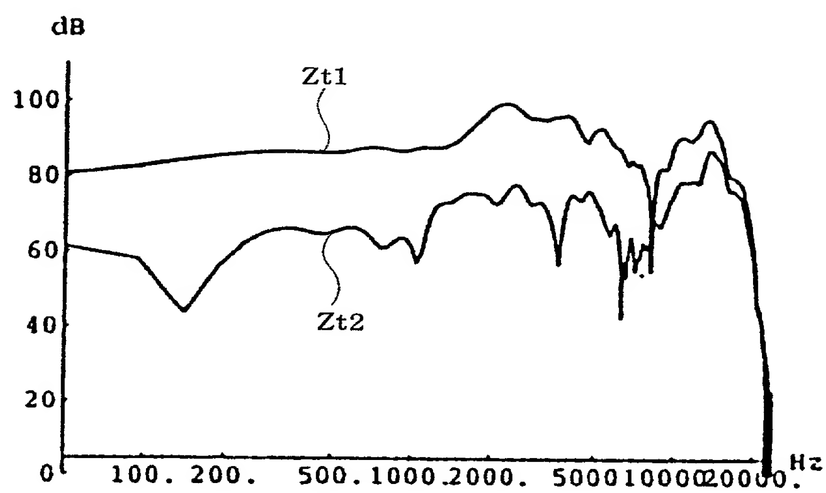

dB
20
10
0
-10
-20
-30
Hdif:96tap
Hsum:32tap
100. 200. 500. 1000. 2000. 5000 10000 20000.
Hz
dB
100
80
60
40
20
0
Zt1
Zt2:ERROR
100. 200. 500. 1000. 2000. 5000 10000 20000.
Hz

FIG.19



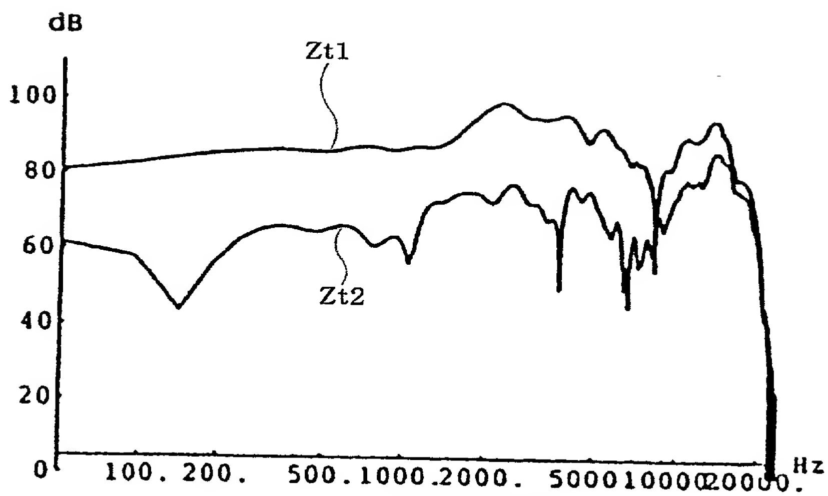
120a
1208
LIN
RIN
sf48k
Hsum
32tap
Z^{-1}
1=7
LOUT
ROUT
205
201
H
2:1
sf24k
HdifA
17tap
sf24k
m=3
Z^{-m}
1:2
H'
202
206
G
sf12k
HdifB
sf12k
n=11:2
Z^{-n}
G'
203
2:1
sf6k
HdifC
1:2
204
sf6k
HdifD
120b

FIG.20



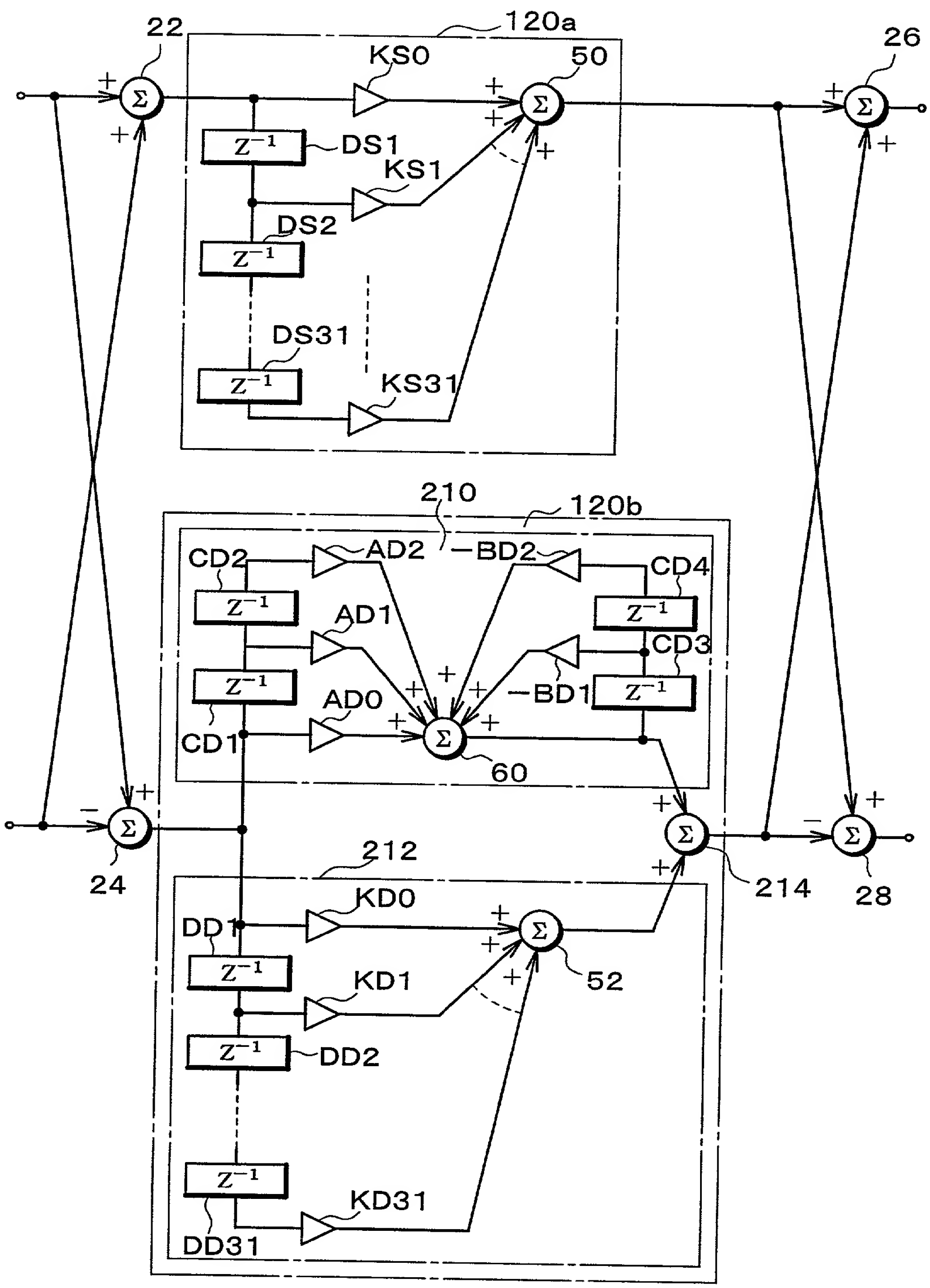
Hsum:32tap
Hdif:128tap

FIG.21



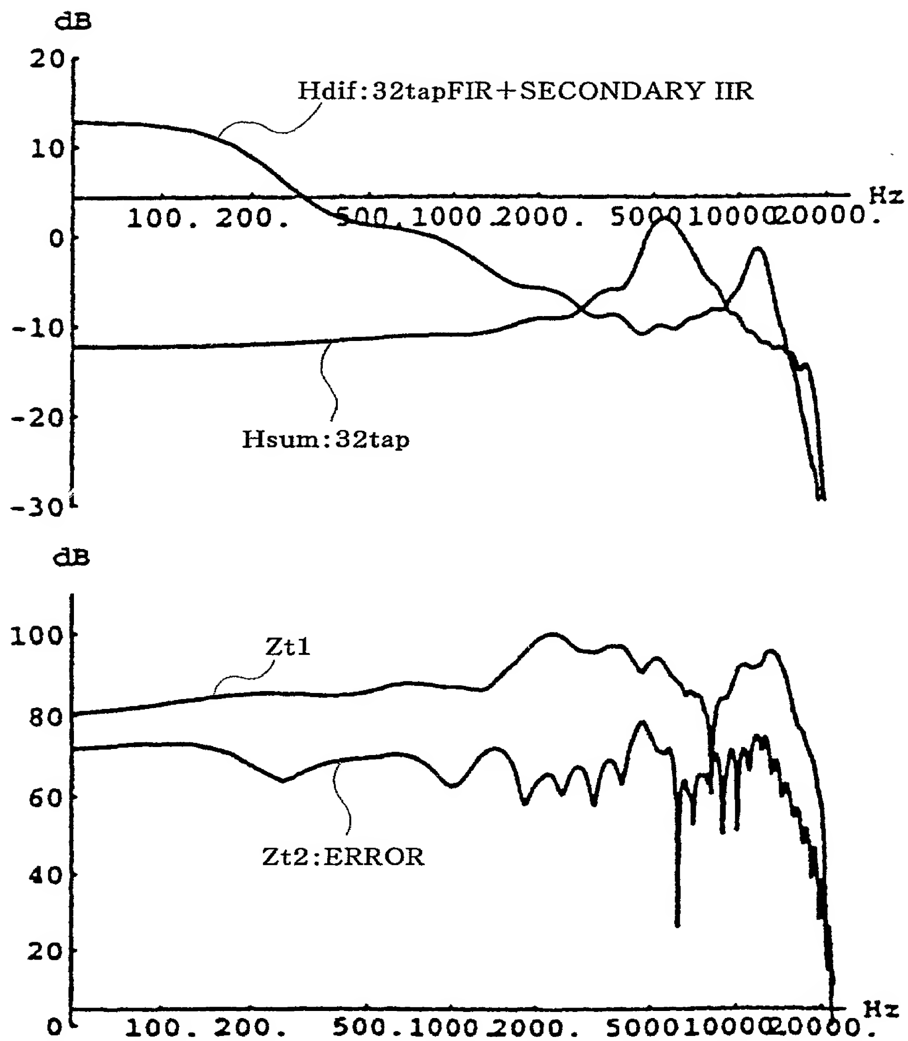
Hsum:32tap
Hdif:FILTER BANK

FIG.22



22
120a
KS0
50
26
DS1
KS1
DS2
DS31
KS31
210
120b
CD2
AD2
−BD2
CD4
AD1
CD3
AD0
−BD1
CD1
60
24
212
214
28
KD0
DD1
52
KD1
DD2
KD31
DD31

FIG.23



dB
20
10
0
-10
-20
-30
Hdif:32tapFIR+SECONDARY IIR
Hsum:32tap
Hz
100. 200. 500. 1000. 2000. 5000 10000 20000.
dB
100
80
60
40
20
0
Zt1
Zt2:ERROR
Hz
100. 200. 500. 1000. 2000. 5000 10000 20000.

FIG.24

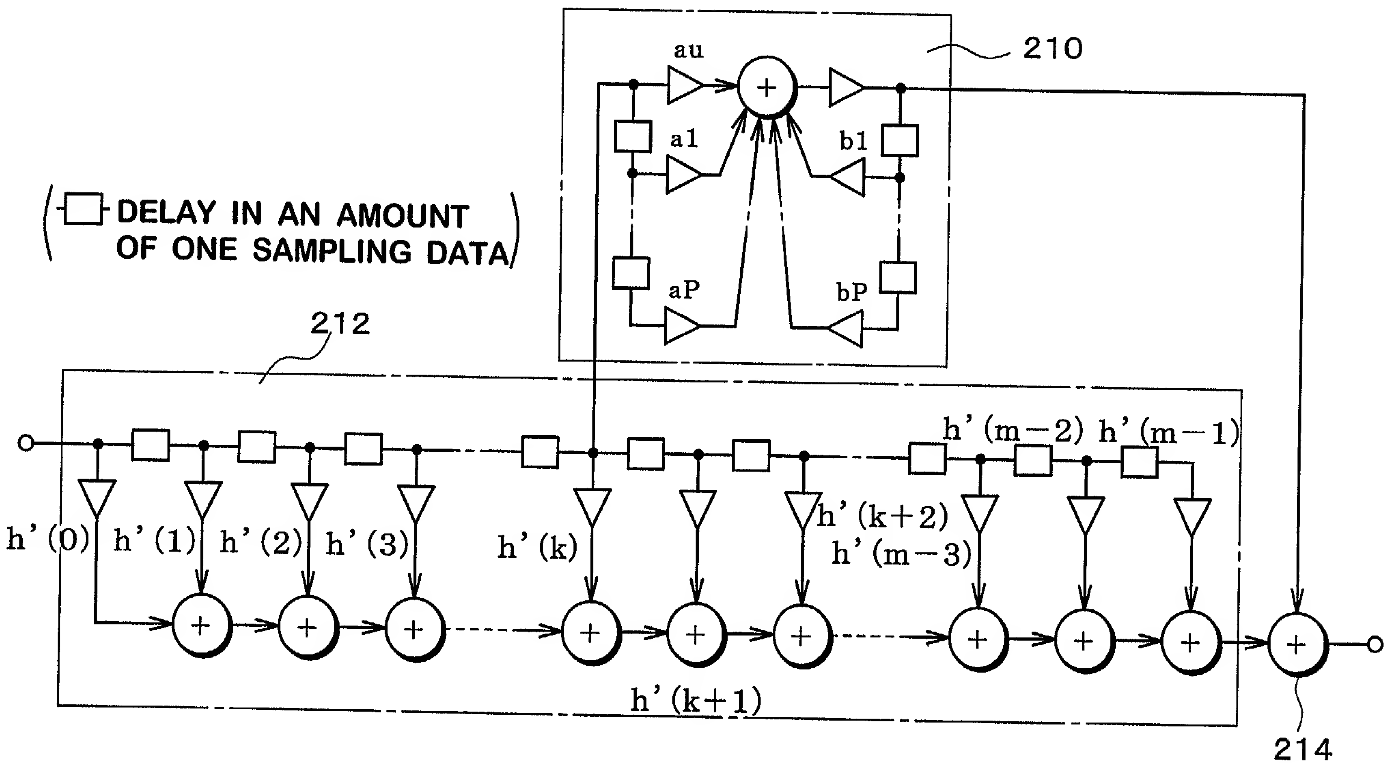
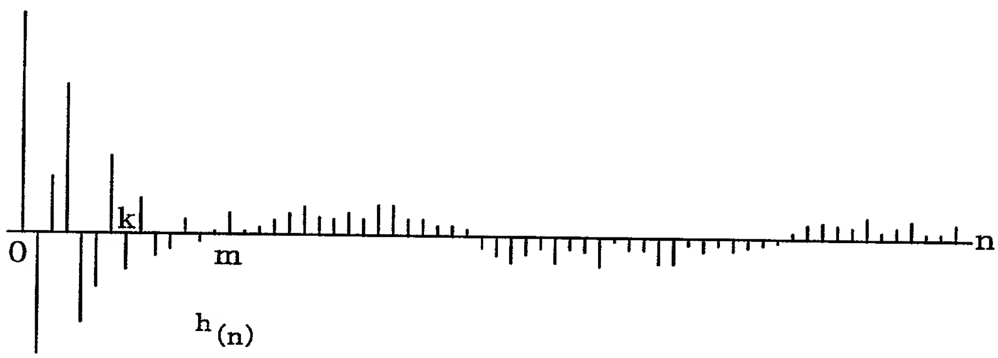
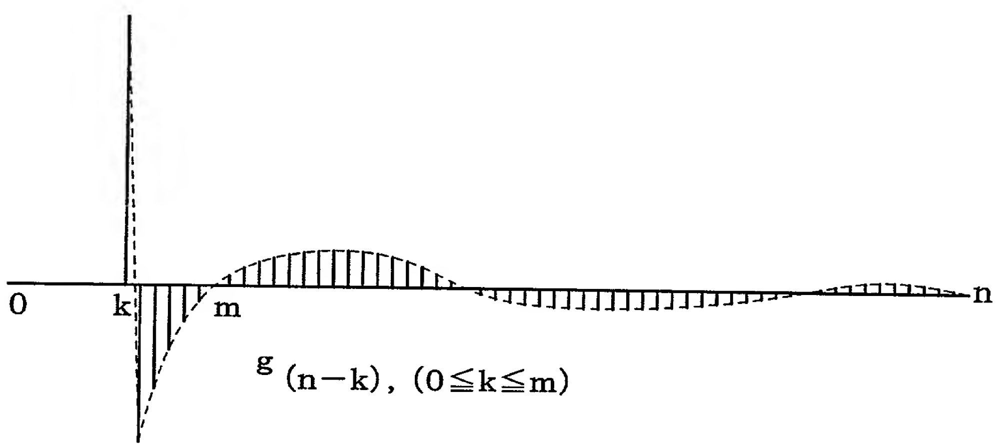

DELAY IN AN AMOUNT OF ONE SAMPLING DATA
210
au
a1
aP
b1
bP
212
h'(0)
h'(1)
h'(2)
h'(3)
h'(k)
h'(k+1)
h'(k+2)
h'(m−3)
h'(m−2)
h'(m−1)
214

FIG.25



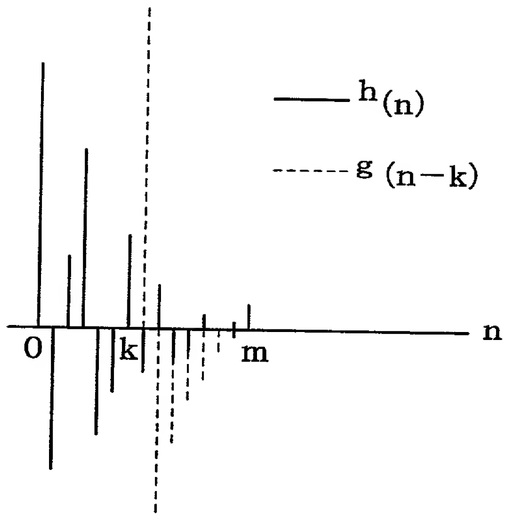
IMPULSE RESPONSE REQUIRED FOR FILTER

FIG.26



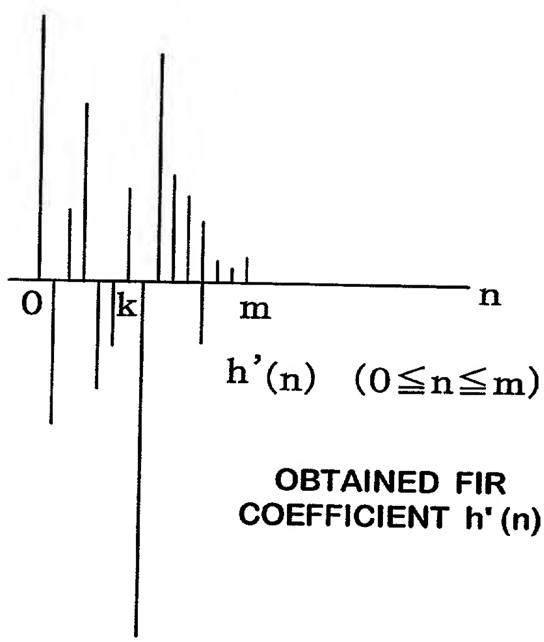
OPTIMUMLY APPROXIMATED IIR FILTER
IMPULSE RESPONSE

FIG.27



COMPARISON BETWEEN h(n) AND g(n-k) IN THE RANGE OF 0≦n≦m

FIG.28



OBTAINED FIR
COEFFICIENT h' (n)

FIG.29

(PRIOR ART)

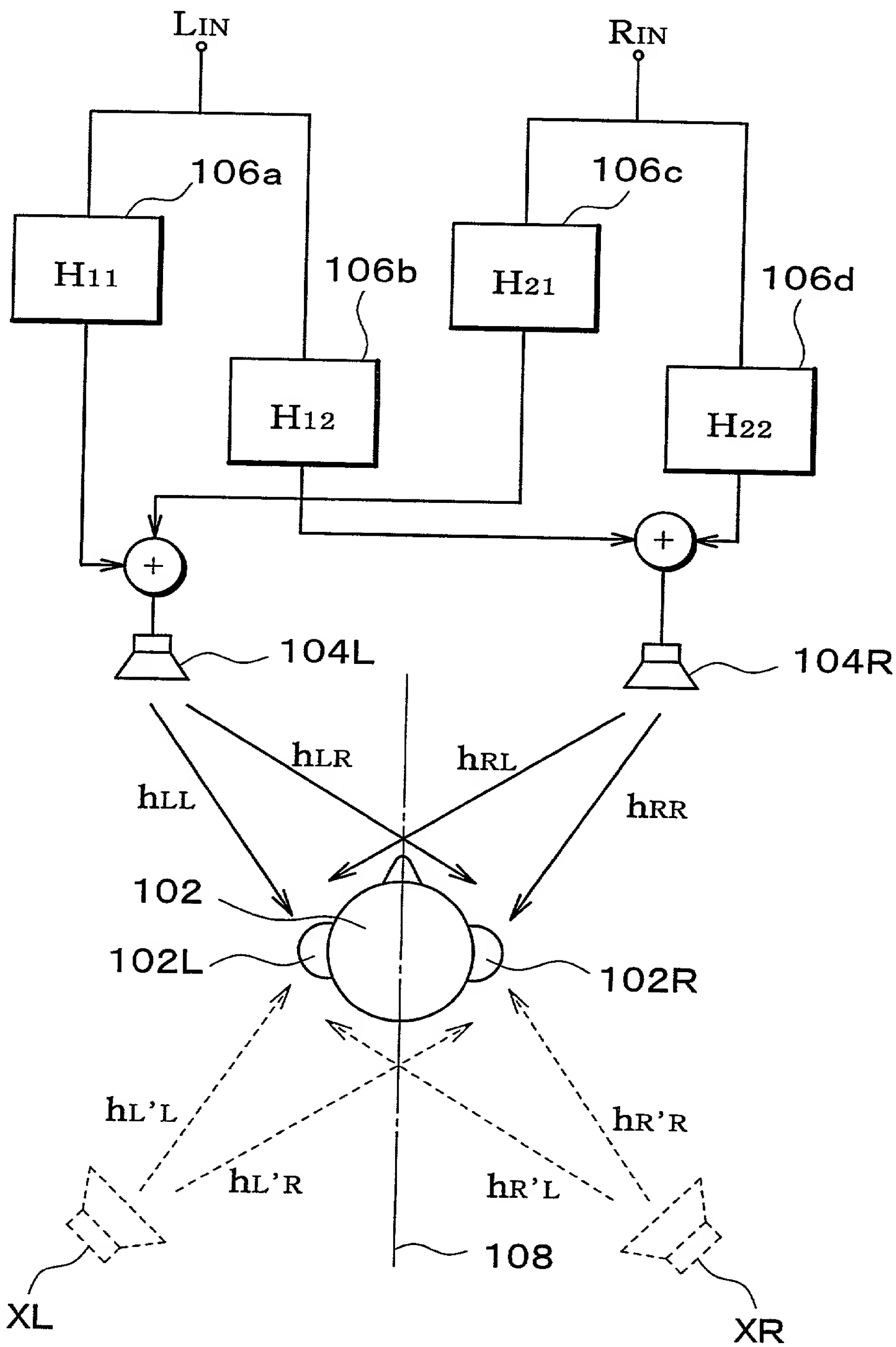


FIG.30

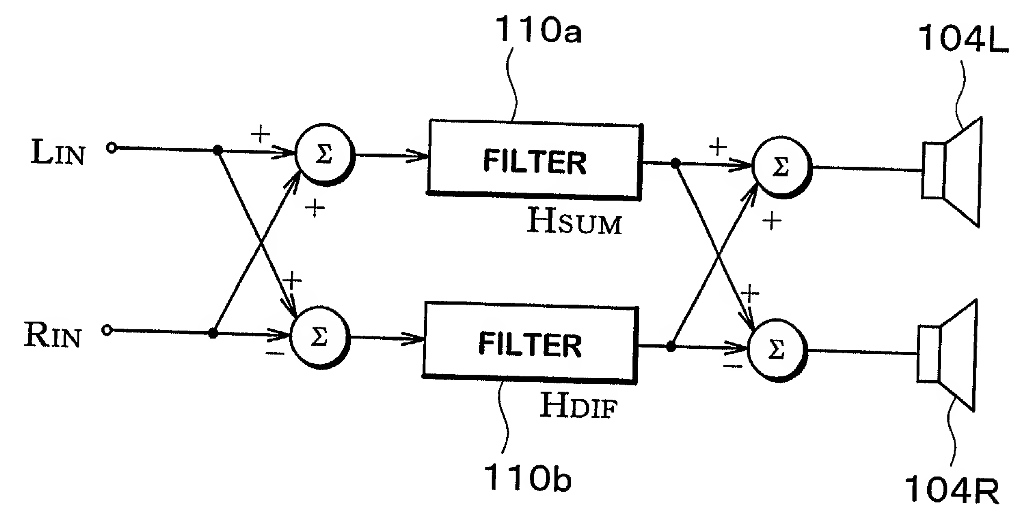
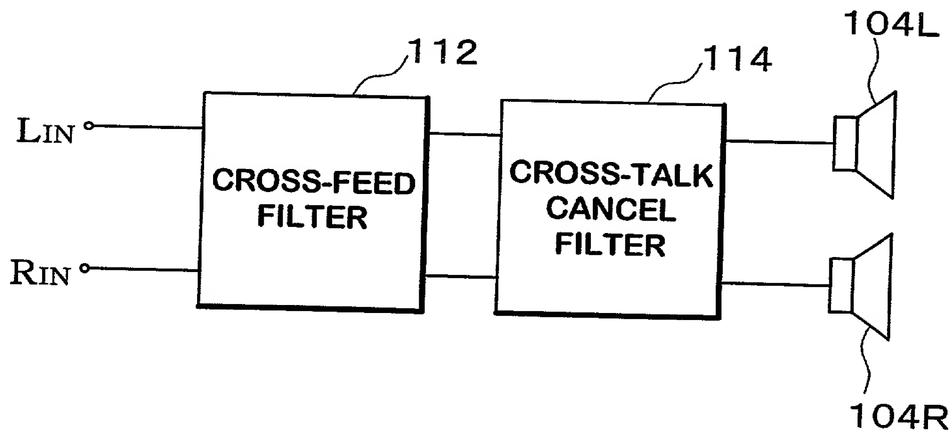


FIG.31



112
114
104L
LIN
RIN
CROSS-FEED FILTER
CROSS-TALK CANCEL FILTER
104R

Docket No.
FUR0007-US

Declaration and Power of Attorney For Patent Application

English Language Declaration

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled

"AUDIO SIGNAL PROCESSING CIRCUIT"

the specification of which

(check one)

☒ is attached hereto.

☐ was filed on ______________ as United States Application No. or PCT International Application Number ______________ and was amended on ______________ (if applicable)

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose to the United States Patent and Trademark Office all information known to me to be material to patentability as defined in Title 37, Code of Federal Regulations, Section 1.56.

I hereby claim foreign priority benefits under Title 35, United States Code, Section 119(a)-(d) or Section 365(b) of any foreign application(s) for patent or inventor's certificate, or Section 365(a) of any PCT International application which designated at least one country other than the United States, listed below and have also identified below, by checking the box, any foreign application for patent or inventor's certificate or PCT International application having a filing date before that of the application on which priority is claimed.

Prior Foreign Application(s)

(Number)	(Country)	(Day/Month/Year Filed)	Priority Not Claimed
10-217929	JAPAN	31 JULY 1998	☐
10-218218	JAPAN	31 JULY 1998	☐
			☐

I hereby claim the benefit under 35 U.S.C. Section 119(e) of any United States provisional application(s) listed below:

(Application Serial No.)	(Filing Date)
(Application Serial No.)	(Filing Date)
(Application Serial No.)	(Filing Date)

I hereby claim the benefit under 35 U. S. C. Section 120 of any United States application(s), or Section 365(c) of any PCT International application designating the United States, listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States or PCT International application in the manner provided by the first paragraph of 35 U.S.C. Section 112, I acknowledge the duty to disclose to the United States Patent and Trademark Office all information known to me to be material to patentability as defined in Title 37, C. F. R., Section 1.56 which became available between the filing date of the prior application and the national or PCT International filing date of this application:

(Application Serial No.)	(Filing Date)	(Status) (patented, pending, abandoned)
(Application Serial No.)	(Filing Date)	(Status) (patented, pending, abandoned)
(Application Serial No.)	(Filing Date)	(Status) (patented, pending, abandoned)

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and transact all business in the Patent and Trademark Office connected therewith. *(list name and registration number)*

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Full name of fourth inventor, if any	
Fourth inventor's signature	Date
Residence	
Citizenship	
Post Office Address	

Full name of fifth inventor, if any	
Fifth inventor's signature	Date
Residence	
Citizenship	
Post Office Address	

Full name of sixth inventor, if any	
Sixth inventor's signature	Date
Residence	
Citizenship	
Post Office Address	